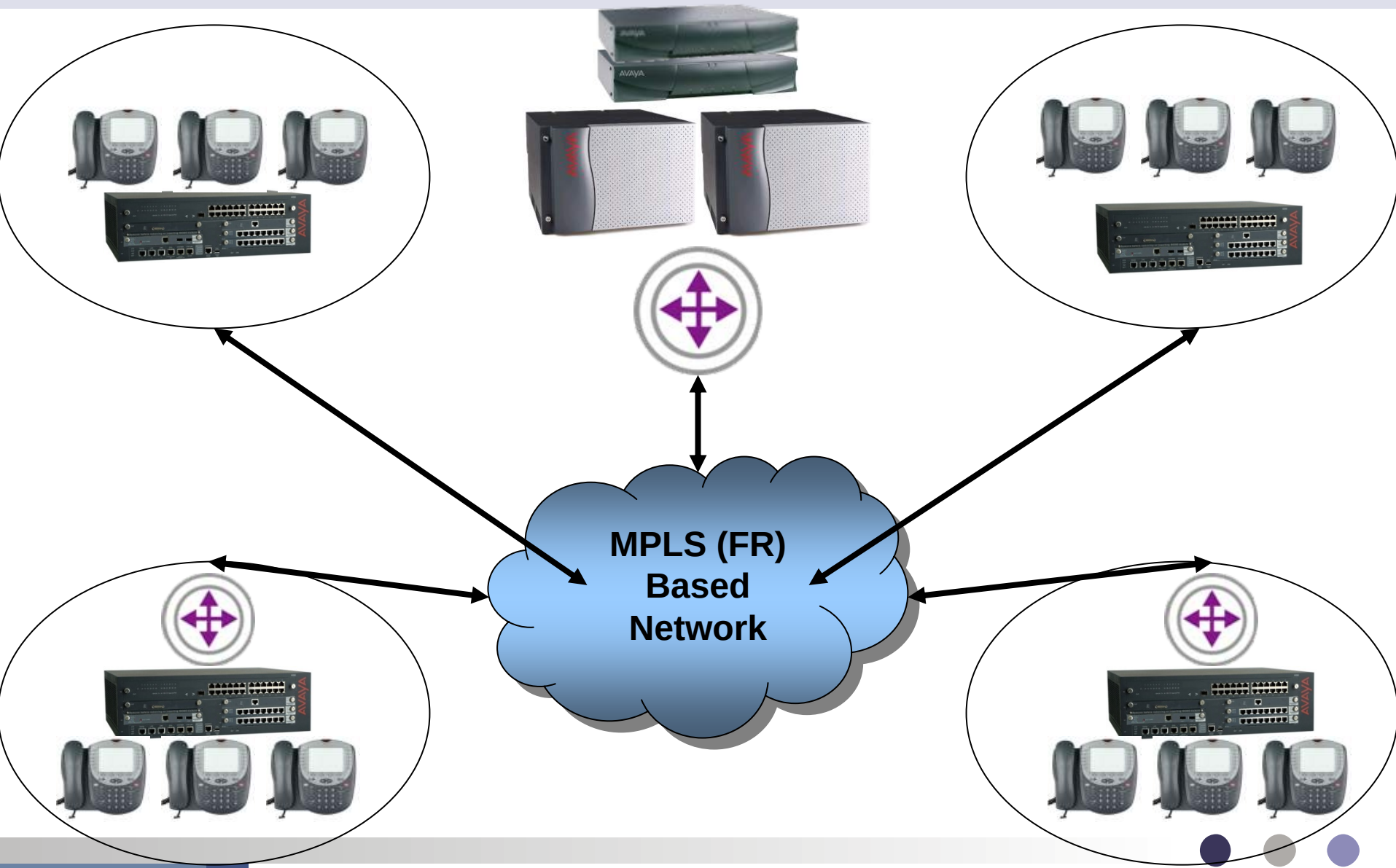
IP Telephony Engineering Principles

Agenda

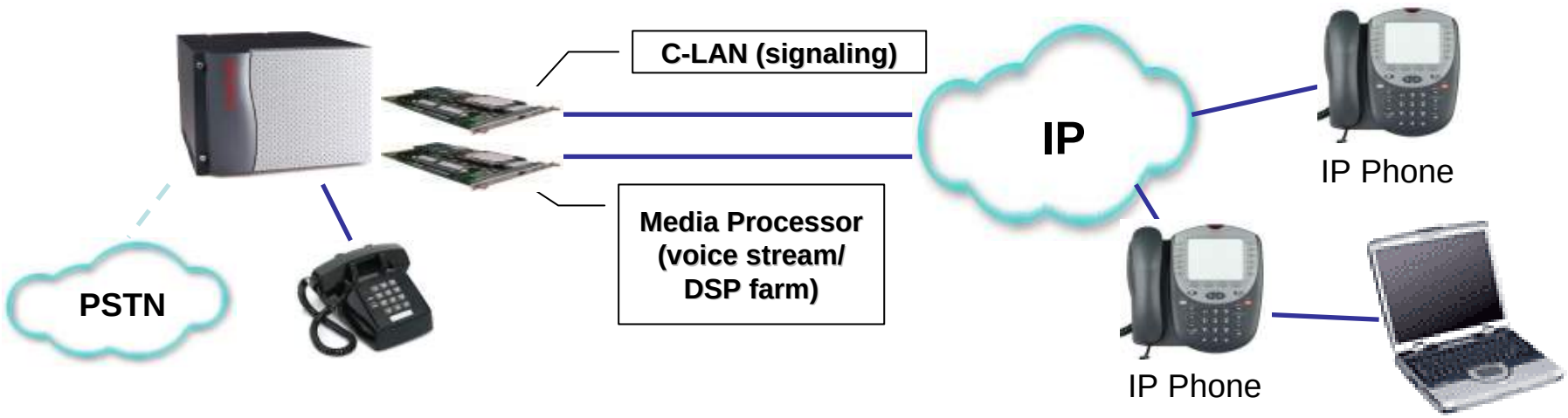
- *Network Region Design*
- *IP Telephone Operation*
- *QOS across the Enterprise*
- *Bandwidth Considerations*
- *Call Admission Control*
- *Media Encryption*
- *IP Trunking & PSTN Fallback*

Note: Discussion will focus on H.323 and not SIP but the concepts with respect to QOS are the same with different ports in some cases.

Typical IP Telephony WAN deployment



Sample IP Connectivity and Functions



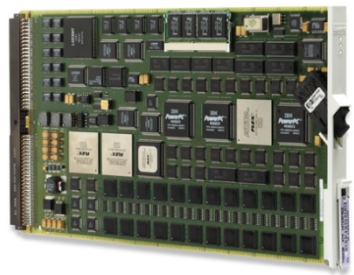
TN Gateway IP Interfaces

IPSI Card: IP Server Interface Card

•Provides Control Interface for MG

•Delivers Tone and Call Classifications Resources

C-LAN: Control LAN card



• Handles signaling for IP Phones and Trunks

• Handles signaling for Adjuncts (Audix, CMS)

•Allows Remote Administration

•Dedicated resource, design to 300 sessions

MedPro Card: Media Processor Card

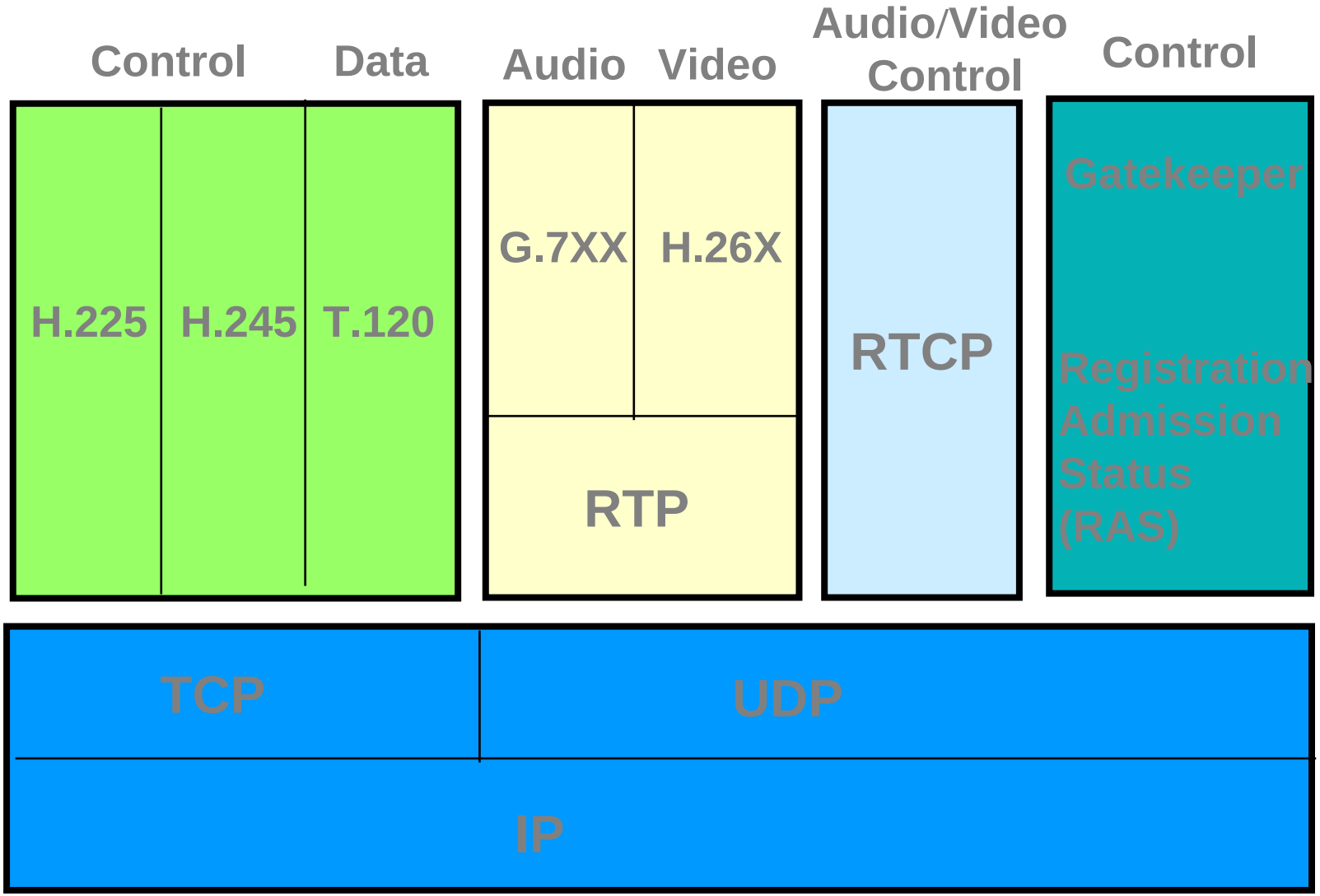
• Converts TDM based media in IP based

• Supports Codecs: G.711, G.729, G.723

• Supports from 32 to 64 simultaneous sessions

• Dynamically allocated resource

H.323v2 Protocol Stack

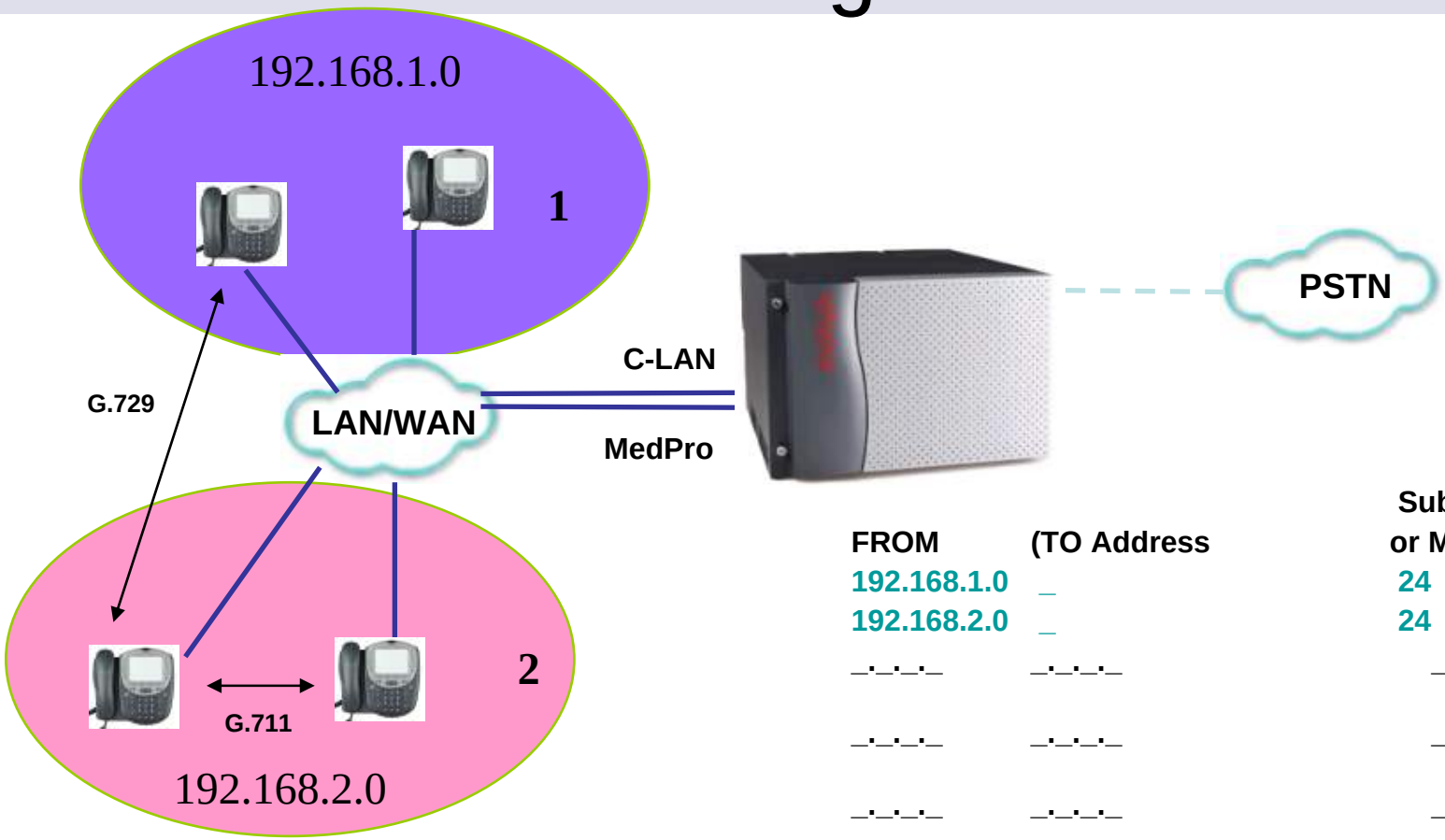


Network Region Design

Network Regions

- Binds Endpoints to a Specific Location
- Dial Plan adjusted by Network Region (useful for E911, local calling)
- Determines what CODEC needs to be used for Intra-Network-Region calls
- Determines what CODEC needs to be used for Inter-Network-Region calls
- Can determine what VoIP Monitor Manager is used
- Determines what QOS settings to be used
  - Customize layer 2, 3, & 4 settings

WAN and Network Regions



FROM	(TO Address	Subnet or Mask)	Region
192.168.1.0	_	24	1
192.168.2.0	_	24	2
_._._._	_._._._	_	
_._._._	_._._._	_	
_._._._	_._._._	_	

DSP Resource Allocation by Call Type

Codec/ Call Type	Capacity Points	Max Calls / Media Processor	
		W/o encryption	W/ AES encryption
G.711 Pass-through Clear Channel	1	64 / (TN2302&MM760) 32 / G350	48/ (TN2302 & MM760) 24/ G350
G.729 & G.723 VoIP	2	32 / (TN2302 & MM760) 16 / G350	24/ (TN2302 & MM760) 12/ G350
Fax Relay Modem Relay T.38	4	16 / (TN2302 & MM760) 8 / G350	Not applicable

Notes: (A) TN2302AP < HV10 (aka TN Media Processor) do not support data transmission other than Avaya patented transport for fax.

(B) A G700 Media Gateway has the equivalent of an MM760 embedded in the system. The G350 has the equivalent of half that.

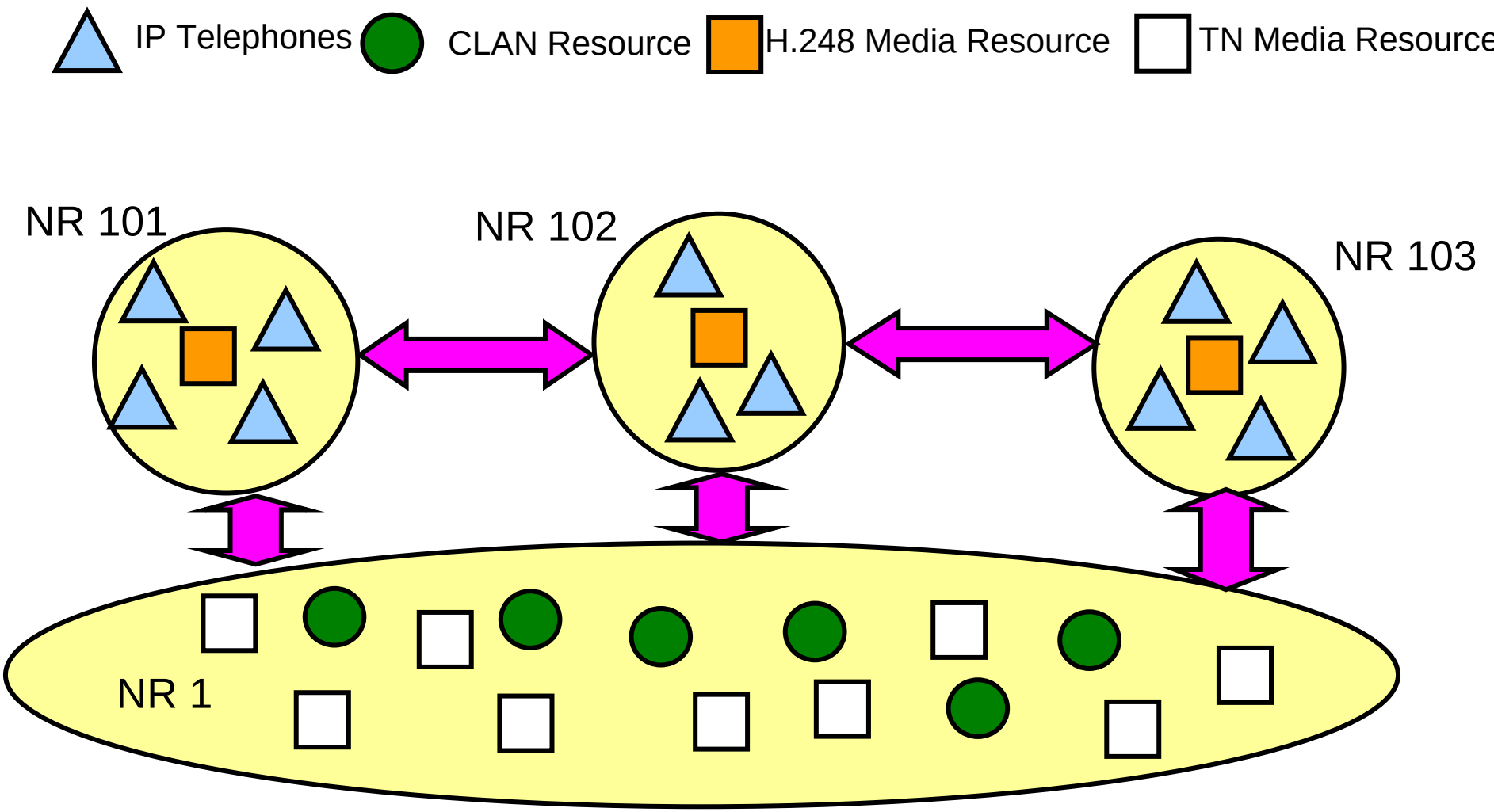
DSP Allocation Rules

1. TN Media Processor already in use by the phone
2. H.248 MG already in use by the phone
3. Preferred region and preferred PN, TN Media Process
4. Preferred region in any PN, TN Media Processor
5. Preferred PN in any region, TN Media Processor
6. Preferred region, H.248 MG
7. Any region, TN Media Processor
8. Any region, H.248 MG

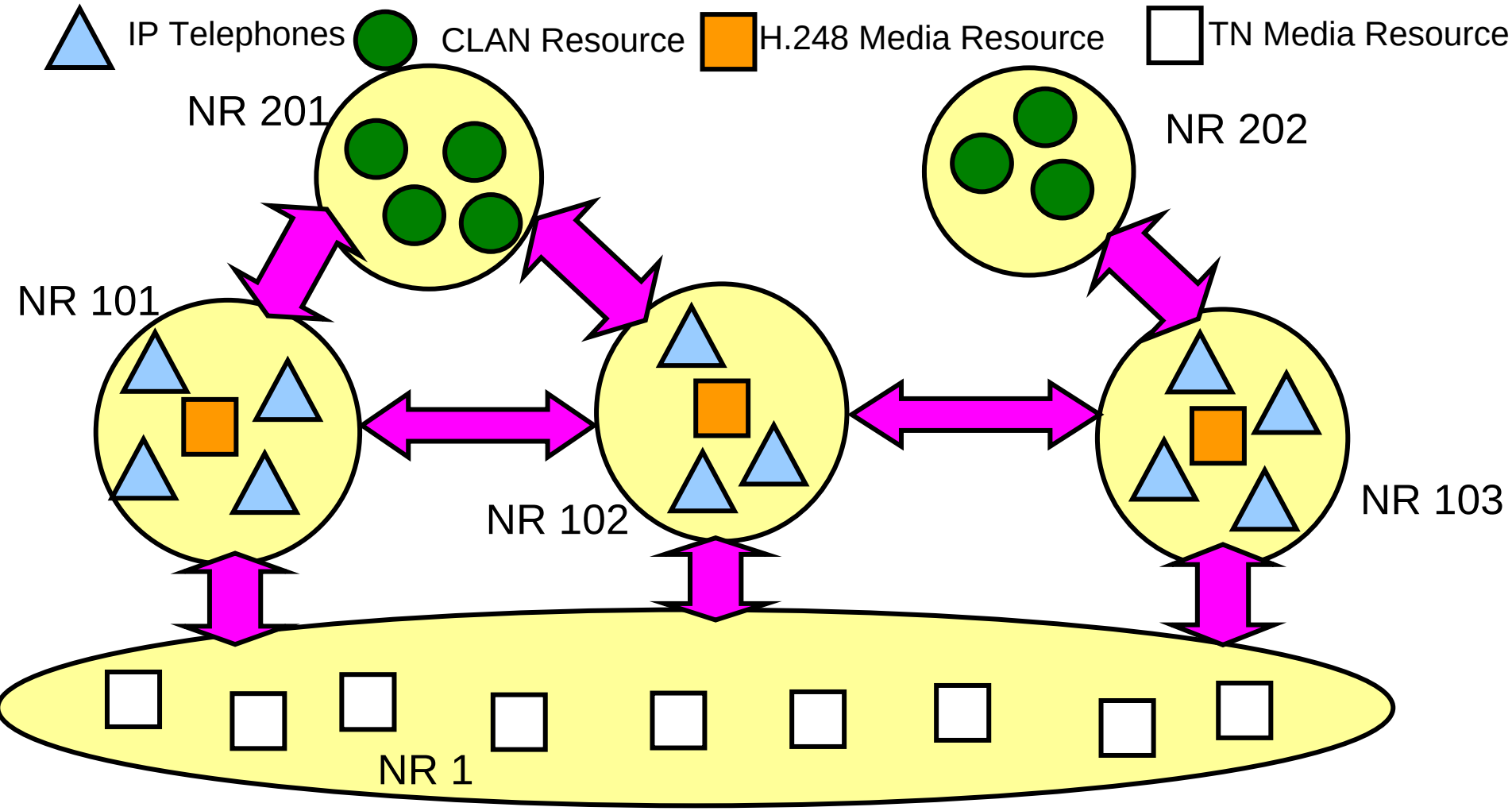
CLAN Design Considerations

- Provide logical mapping from IP Telephone NR to CLAN pool regions
- Maximum of 300 registered endpoints per CLAN even in failover scenario (N+1)
- Have at least 2 CLANs for MGC list; for max CLAN resiliency use 3
- Maximize operational efficiency by minimizing the number of locations in each pool
- Keep it simple or make it manageable

Designing for CLANs By locations



Designing for CLANs -- Logical Pooling



CLAN Pooling

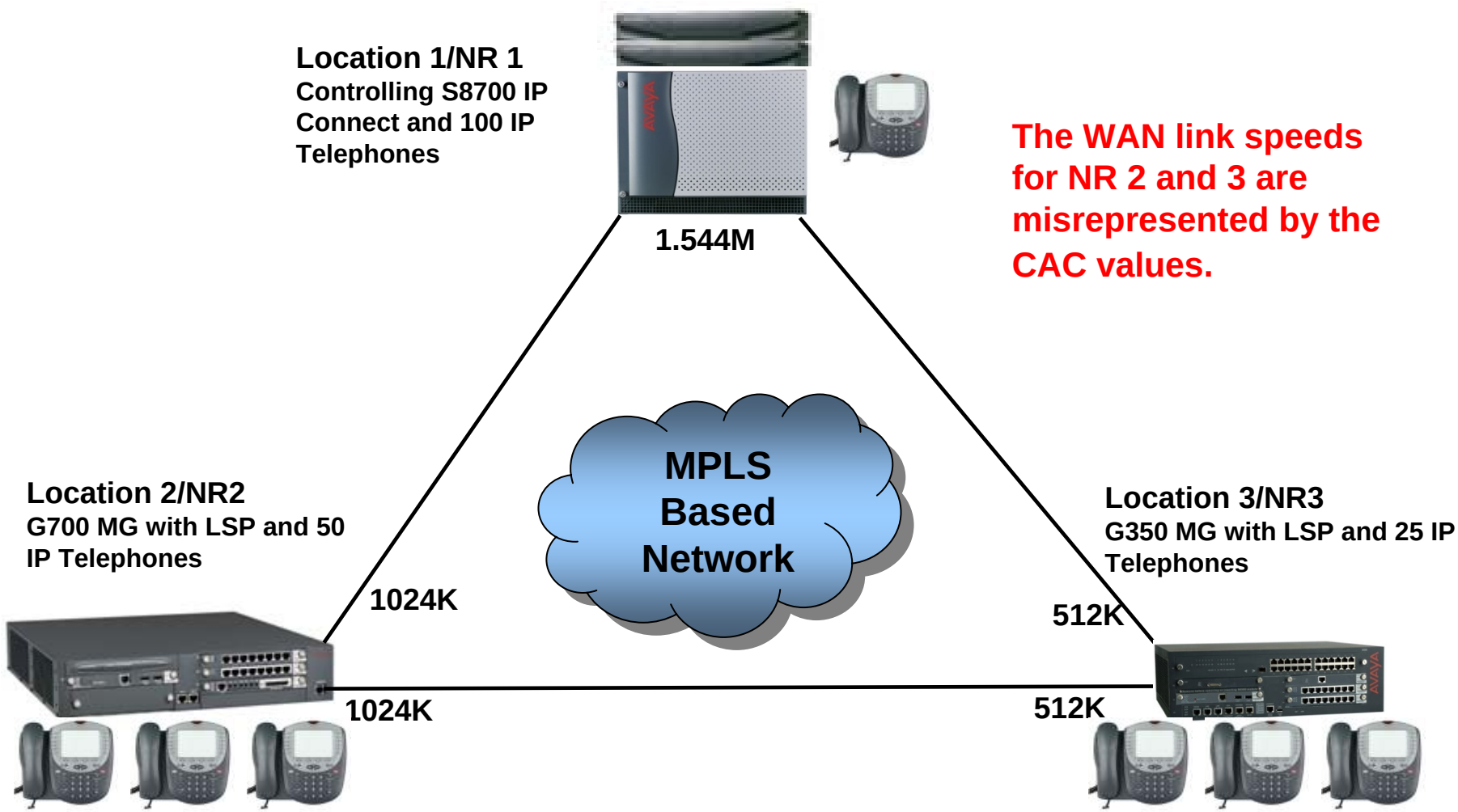
Benefits

- More Granular Registration Control
- Better Trouble Isolation
- Better recovery control
- Greater flexibility in the application of network policy

Negatives

- CLANs registrations will not be balanced across network regions
- Need more CLANS
- Greater operational complexity
- Operational changes may require re-design

NR Design without "Ghost" Region


Location 1/NR 1
Controlling S8700 IP Connect and 100 IP Telephones
1.544M
The WAN link speeds for NR 2 and 3 are misrepresented by the CAC values.
MPLS Based Network
Location 2/NR2
G700 MG with LSP and 50 IP Telephones
1024K
1024K
Location 3/NR3
G350 MG with LSP and 25 IP Telephones
512K
512K

“Ghost Regions”

- In order to correctly define the WAN link for each site, a “Ghost Region” is configured so the CAC values are correct
  - All 3 of our Network Regions in the previous example would directly connect to the Ghost Region
    - The interconnection from NR 1 to NR 2 would intervene through the Ghost Region
  - By using the Ghost Region configuration, the CAC bandwidth limits would be correctly defined for the actual WAN link and prevent over subscription

NR Design with“Ghost” Region

Location 1/NR 1
Controlling S8700 IP Connect and 100 IP Telephones

1.544M

MPLS

NR 5

Location 2/NR2
G700 MG with LSP and 50 IP Telephones

1024K

512K

Location 3/NR3
G350 MG with LSP and 25 IP Telephones

The MPLS WAN is now represented by NR 5 in Communication Manager and the topology is correctly depicted by the CAC values

Administration of GR

```
change ip-network-region 1                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                          Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions        Gateway
 1   1     1
 1   2     2      n                     5  :  :  :
 1   3     2      n                     5  :  :  :
 1   4
 1   5     2      y       1544:Kbits
 1   6
```

```
change ip-network-region 2                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                          Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions        Gateway
 2   1     2      n                     5:  :  :
 2   2     1
 2   3     2      n                     5:  :  :
 2   4
 2   5     2      y       1024:Kbits
 2   6
```

```
change ip-network-region 3                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                          Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions        Gateway
 3   1     2      n                     5:  :  :
 3   2     2      n                     5:  :  :
 3   3     1
 3   4
 3   5     2      y        512:Kbits
 3   6
```

```
change ip-network-region 5                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                          Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions        Gateway
 5   1     2      y       1544:Kbits
 5   2     2      y       1024:Kbits
 5   3     2      y        512:Kbits
 5   4
 5   5     1
 5   6
```

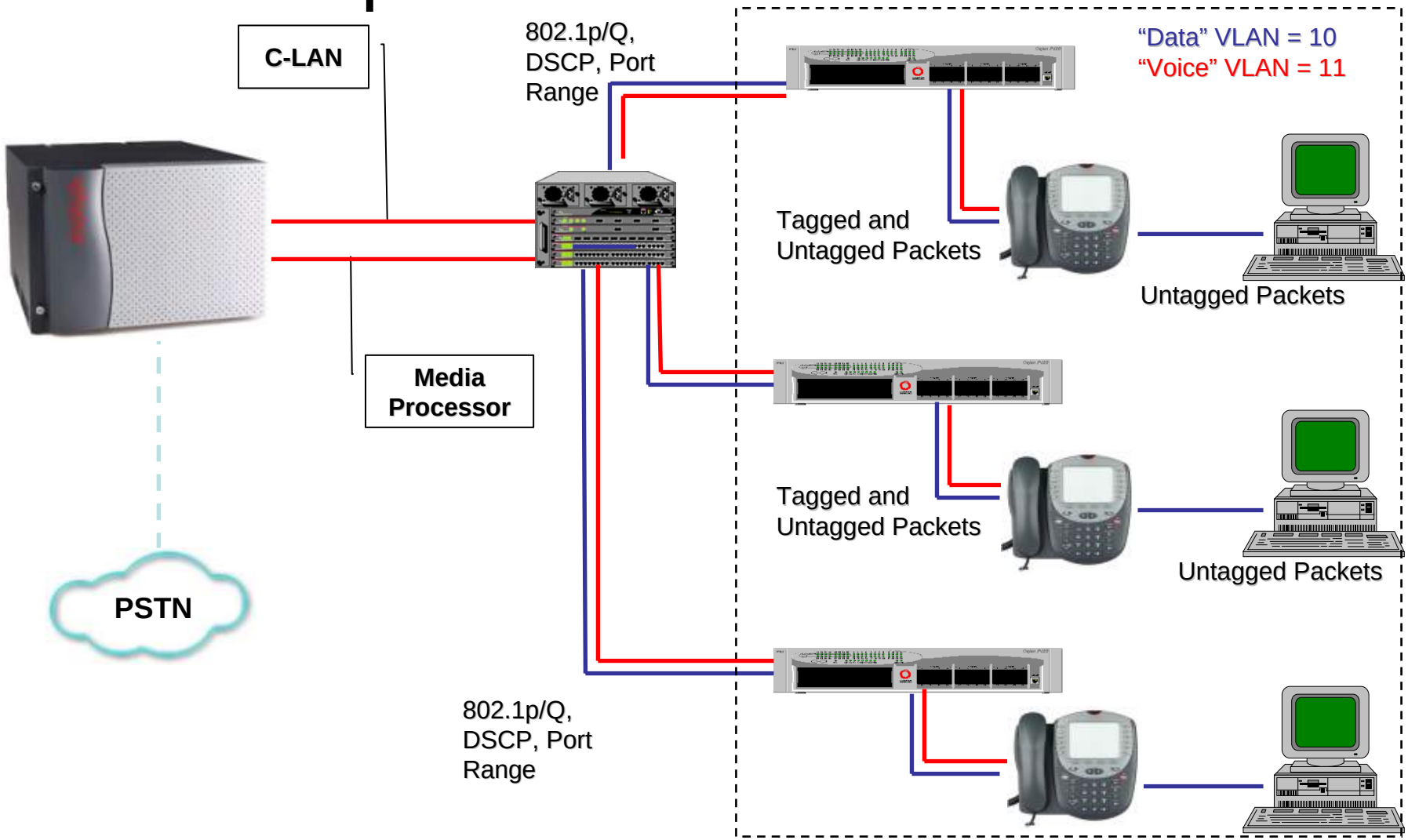
Network Region 5 is used as the Ghost Region (for actual implementations, a higher region may be more appropriate to allow for scalability).

The only region that NR's 1-3 directly connect to is NR 5 and intervene to the other regions. There are no actual resources in NR5, it is more of the WAN cloud representation.

All WAN connections are now correctly defined and Communication Manager has the correct interpretation of the topology.

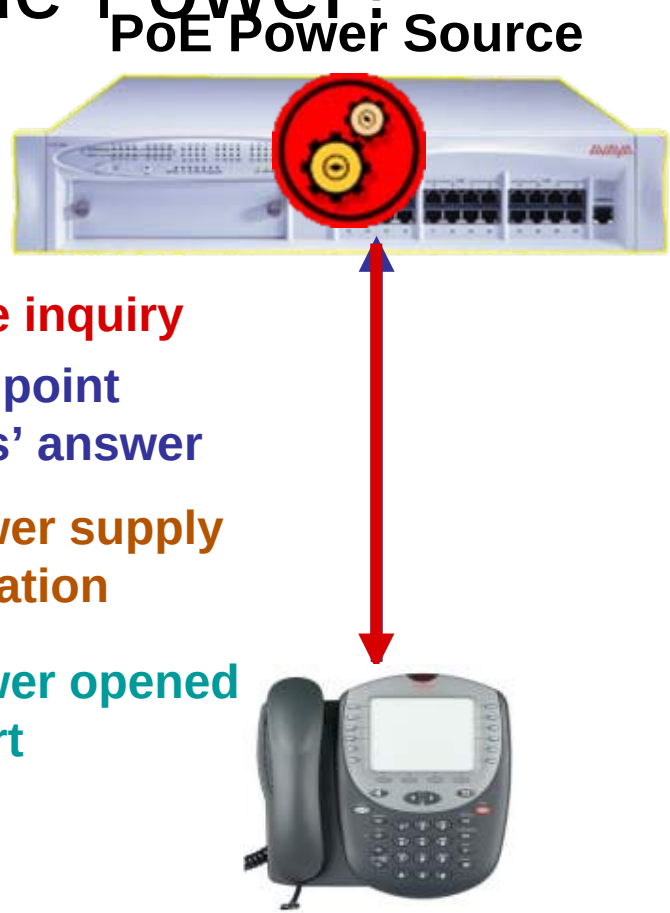
IP Telephone Operation

Implementation Overview



Power over Ethernet – How does it Provide Power?

- IP Phones have been 802.3af compliant for years
- Performs phone detection
- It applies power to the endpoint (IP phone) using the signaling pairs
- If the endpoint is removed or the link is interrupted
  - Power is shut off
  - the detection process starts again



Power Consumption

Watts (IEEE 802.3af -2003@ 48V)

Class 2

Class 3

	4601/4602	4602SW	4610SW	4620	4620SW	4620SW	4621SW/ 4622SW	4625SW	4630SW
Typical	3.5	4.1	4.0	7.7	5.9	4.6	4.9	7.8	11.8
Worst Case	4.6	5.0	6.0	9.9	8.0	5.75	6.45	9.42	12.9

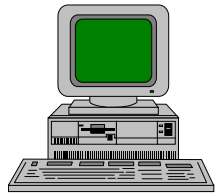
Typical is measured off-hook. Worst Case is analytical. Except the 4601 and 4602 all telephones had a PC attached at 100Mbps. The EU24 adds less than 1W to the 4620 and 4620SW numbers. The EU24BL can not be used with POE, use the 1151B. The 4620SW CR can be identified by the ethernet jacks that point down, rather than directly back out of telephone.

DHCP Process – Dual VLAN

User PC

DHCP Discover

Offer IP address in VLAN 10

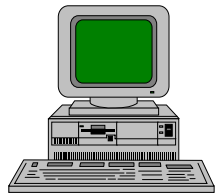


DHCP Server

Offers:
-IP ADDR
-Subnet Mask
-Default Gateway

DHCP Discover

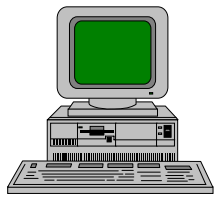
Offer IP address in VLAN 10



DHCP Server

Offers:
-IP ADDR
-Subnet Mask
-Default Gateway
-Site Specific Option (176):
-GateKeeper IP Addr (8)
-GateKeeper Port
-QoS Parameters:
-802.1Q = 1
-VLAN = 11
-802.1p = 6
-TFTP Address (8)

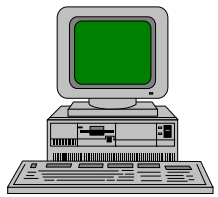
DHCP Release



DHCP Server

DHCP Discover Using VLAN Tagging (11)

Offer IP address in VLAN 11



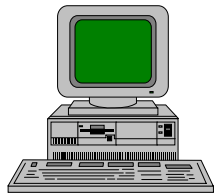
DHCP Server

Once the phone knows what the voice VLAN is, it will boot into that VLAN first

IP Phone registration Process

DHCP Discover

Offer

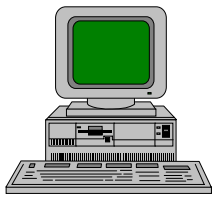


DHCP Server

Offers:
-IP ADDR
-Subnet Mask
-Default Gateway
-Site Specific Option:
-GateKeeper IP Addr (8)
-GateKeeper Port
-QoS Parameters
-TFTP Address (8)

TFTP Get

TFTP Put



TFTP Server

TFTP Puts:
-Boot Code (First Time)
-Application Code (First Time)
-Config (e.g. QoS)

Enter Extension
Enter Password

Registration, Admission, Status

H.323 and Feature Functionality

Validates:
-Extension
-Password

Provides:
-Access to medias
-Feature / Functionality

DHCP Considerations

- Telephone Firmware 2.1+ offers optional use of IP address after lease expires
  - Does not protect against power failure or reboot
  - Based on administrable DHCPSTD parameter
  - DHCPSTD = 0 means “Despite the DHCP Standard, continue using the current IP address after the lease expires, but:
    - Send DHCPREQUEST about every minute
    - Send ARPREQUEST every 5 seconds
    - If ARP REPLY received, set IPADD to 0.0.0.0 and re-initiate DHCP Discovery”
  - DHCPSTD = 1 means “Follow the DHCP Standard (RFC 2131, Section 4.4.5); give up the IP address immediately if DHCP lease expires”

DHCP Considerations

- Each subnet requires a DHCP scope.
- For clients not on the same subnet as the DHCP server, enable DHCP relay on the router interface for the client subnet (i.e. “ip helper-address”)
- Embedded DHCP server within G350 to support IP phones and local IP stations
- No plans to support G700 DHCP
- Can use local router as a DHCP server

```
service dhcp
ip dhcp pool “Miami branch office"
network 10.10.10.0 255.255.255.0
default-router 10.10.10.1
lease 120
option 176 ascii MCIPADD=X.X.X.X,MCPORT=1719,TFTPSRVR=X.X.X.X
ip dhcp excluded-address  10.10.10.1
```

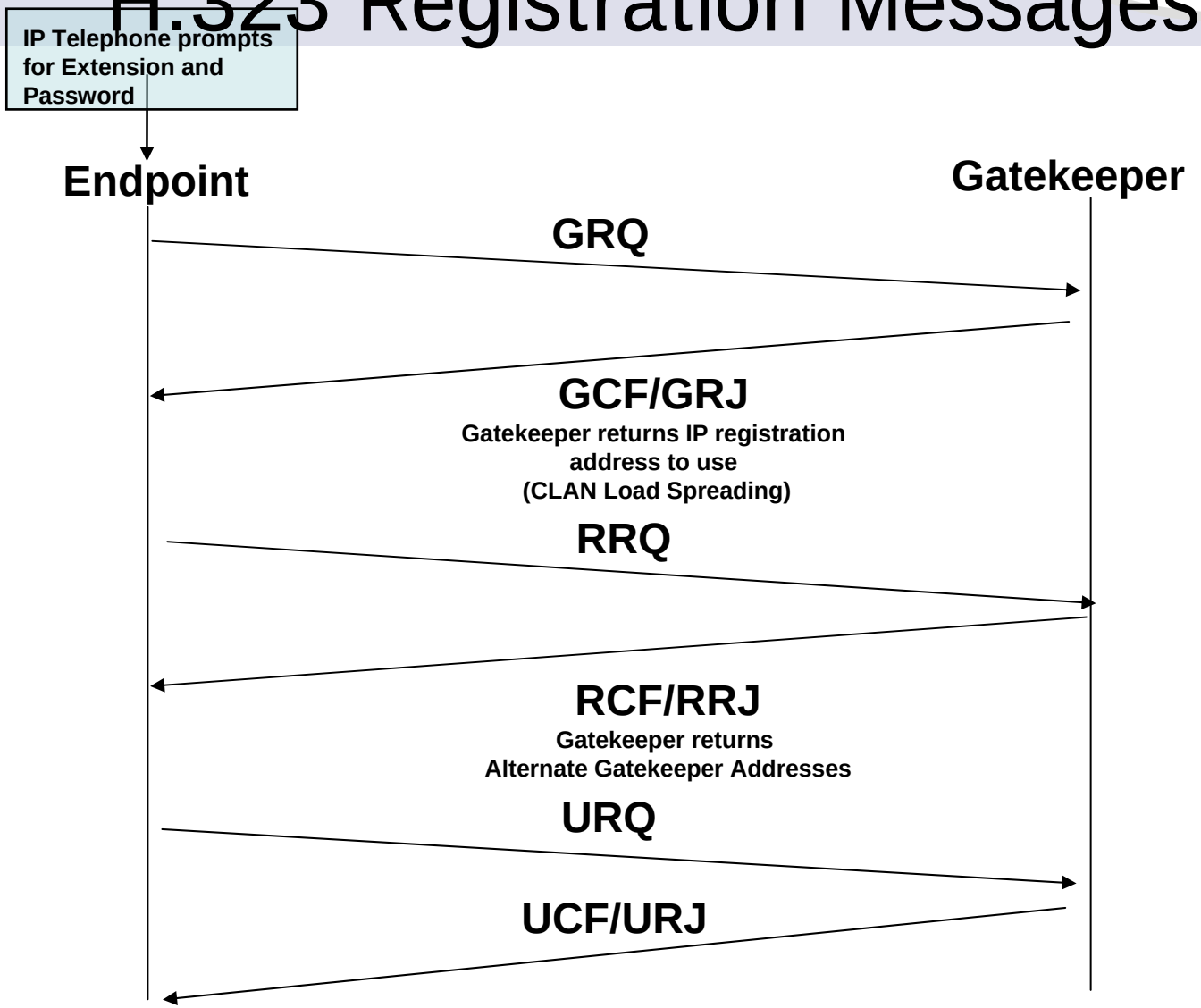
TFTP Server

- Used for upgrades and optional configuration files
- Not a point of failure for basic telephony operation
- Possible point of failure for additional features
  - Review each configuration option to be used in order to determine impact of failure
- Embedded TFTP server within G250/G350 to optimize local IP-phones upgrade process
  - Limited space in NVRAM

What happens during Registration?

- Registration starts, GRQ, GCF, RRQ
- Phone asking user for Login (extension) and password
- Phones sends request for registration for the extension
- Server sends to the phone an encrypted message to validate password
- Password is validated, Server sends a RCF, and features to the phone and all relevant timers
- Phone sends the supported CODECS and other relevant parameters
- Based on the C-LAN or the Phone's IP address, it is set to a specific Network Region

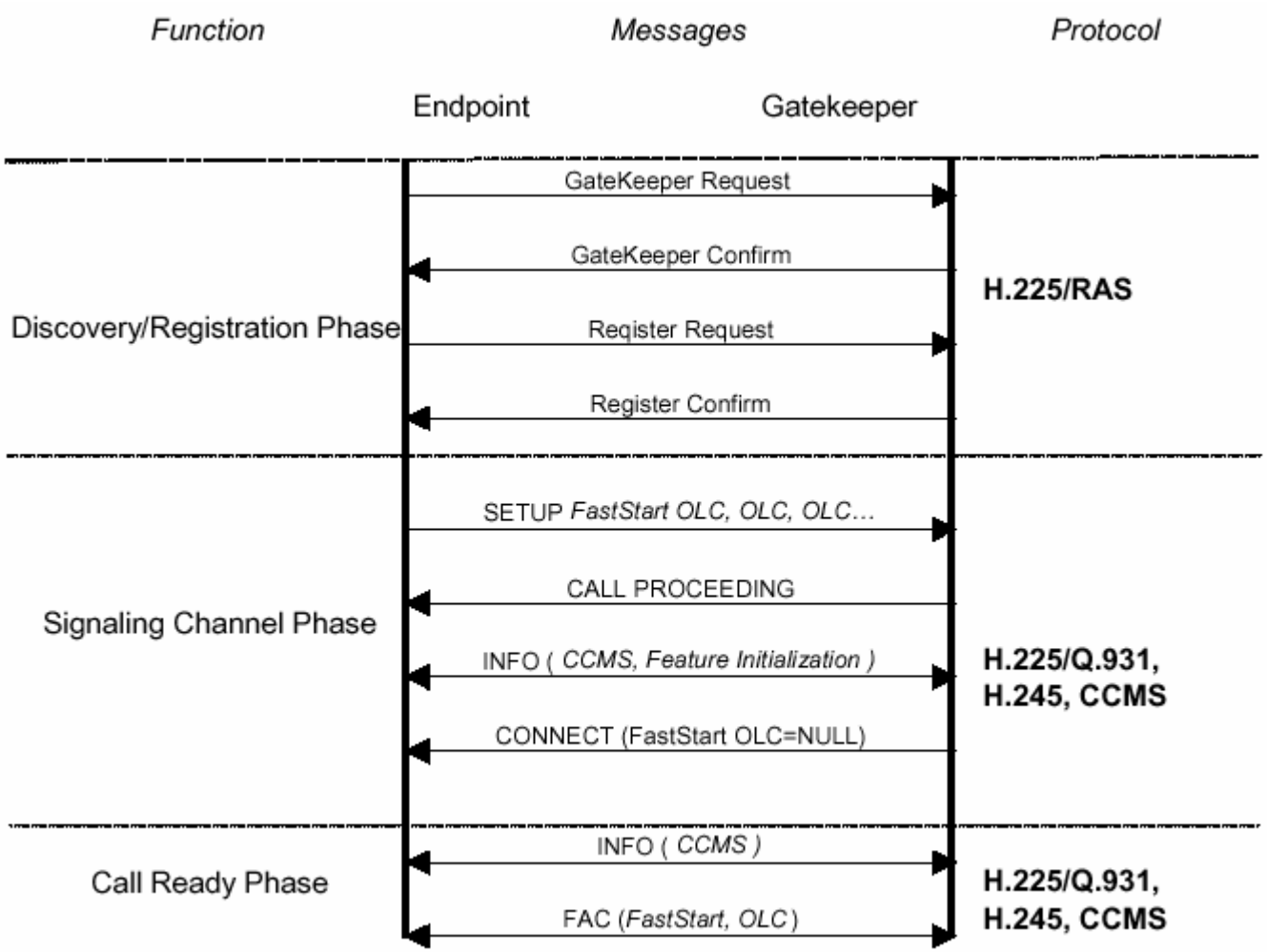
H.323 Registration Messages



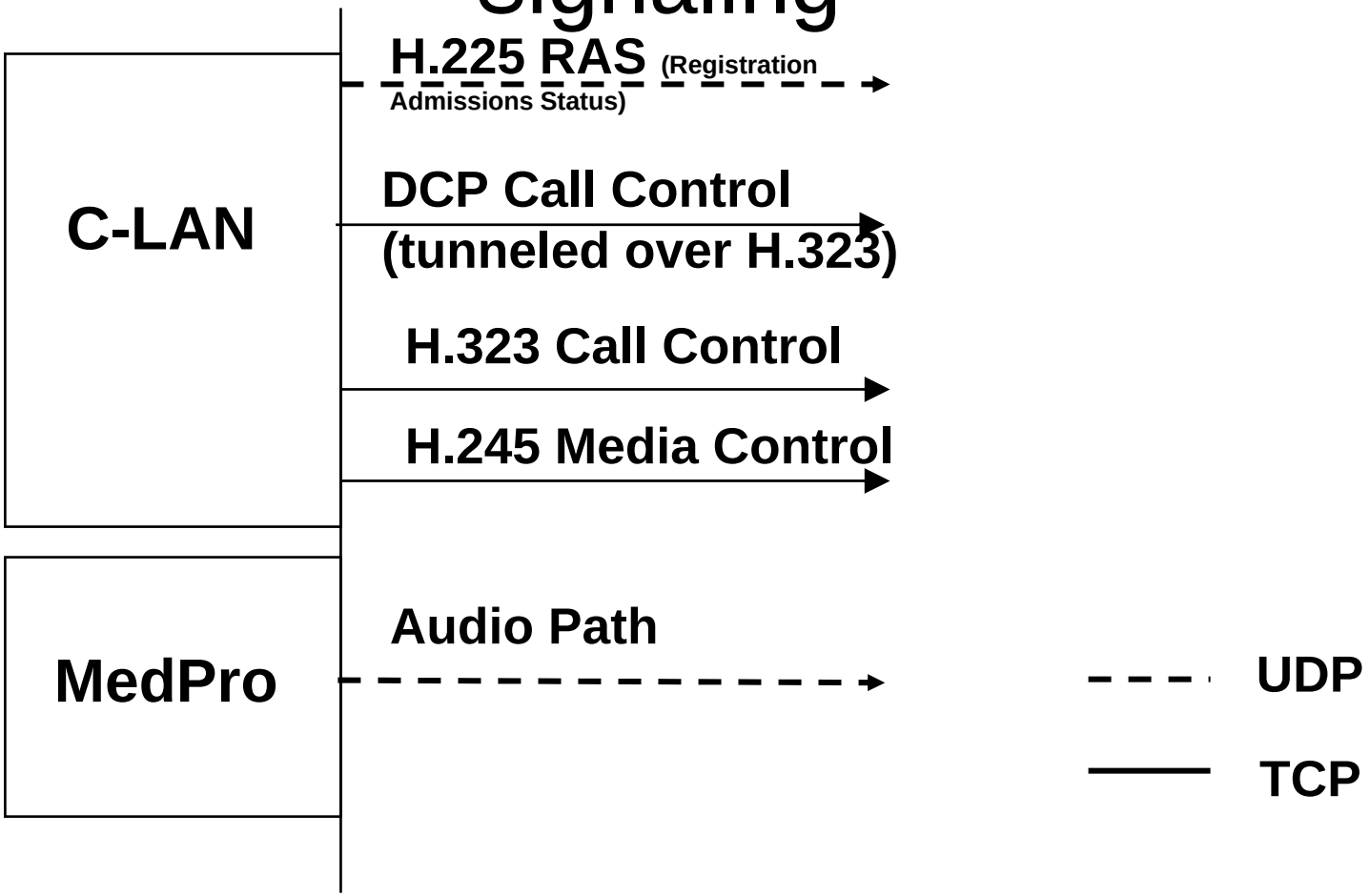
CLAN Load Spreading

- Communication Manager sends as the RAS address in the GCF the IP address of a CLAN in the same network region as the CLAN that received the GRQ.
- Communication Manager software will select the registration address in a cyclical fashion.
- Use this ability to balance registration across multiple CLANS.
- IP endpoints will accept an address in the GCF and use it for that registration.
- Balancing only occurs during registration; phone does not change CLANs during normal operaion

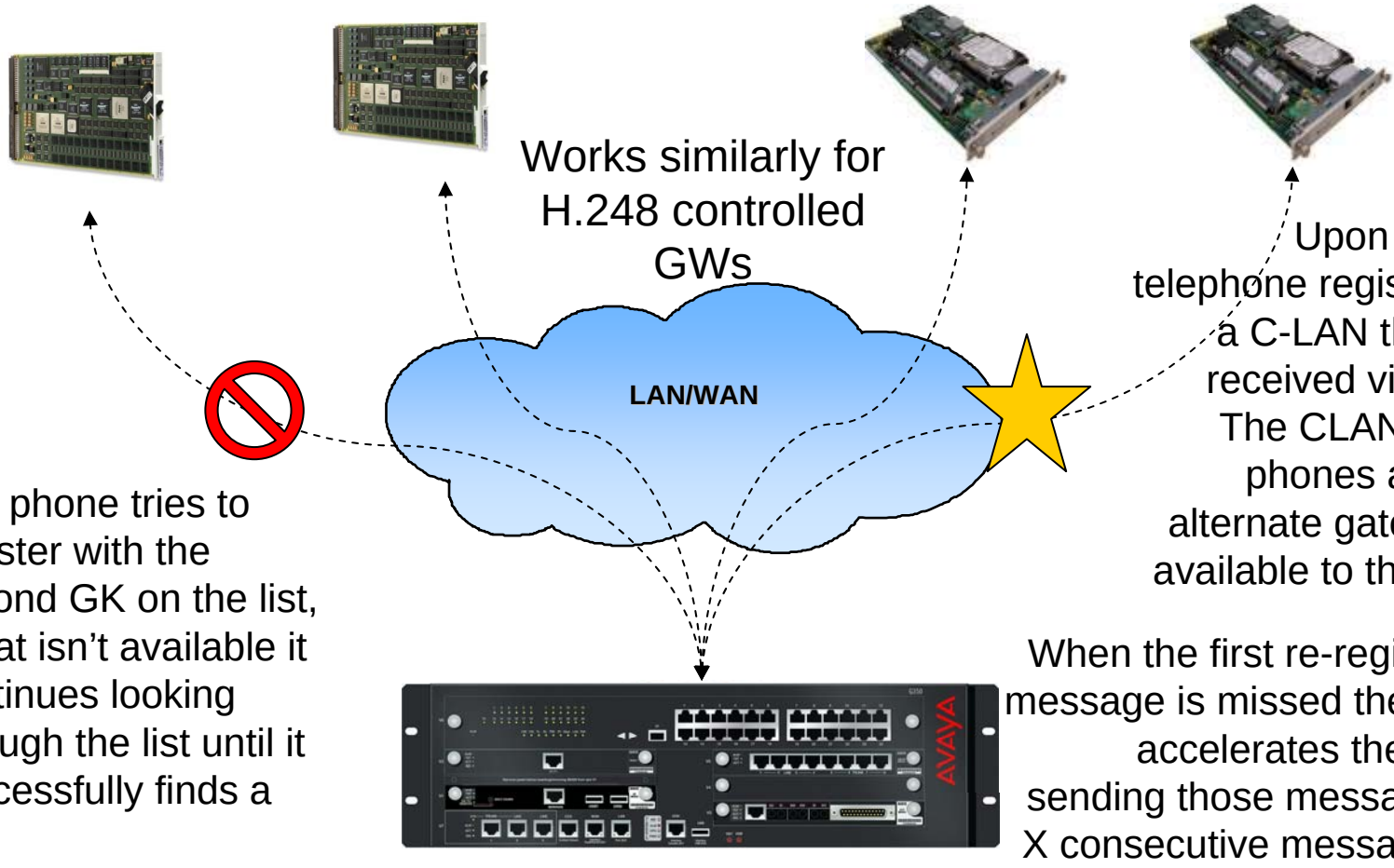
H.323 Signaling Messaging



Communication Manager -- IP Station Signaling



Alternate Gatekeeper



Works similarly for H.248 controlled GWs

Upon first boot telephone registers with a C-LAN that it has received via DHCP. The CLAN tells the phones about the alternate gatekeepers available to the phone.

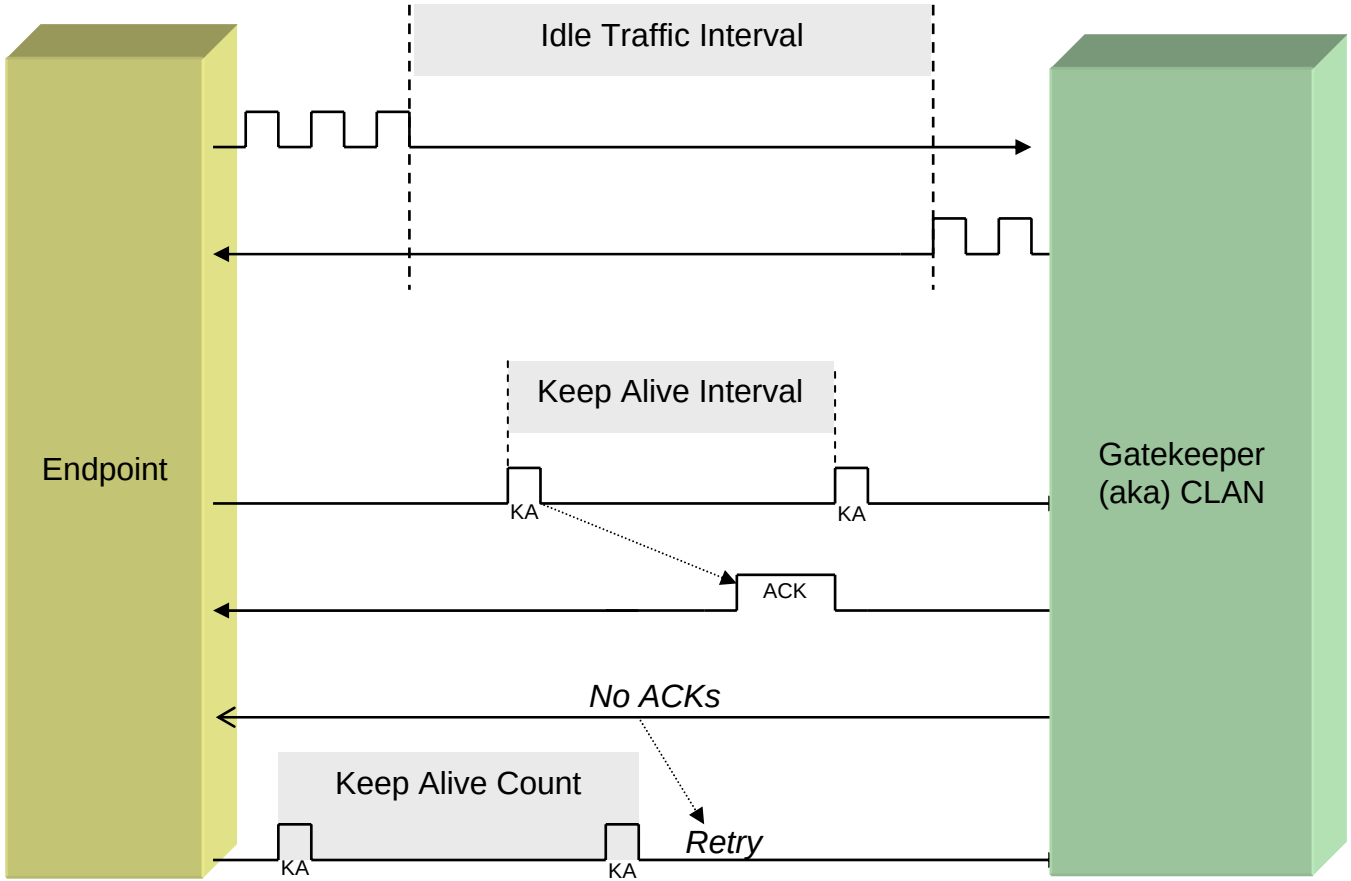
The phone tries to register with the second GK on the list, if that isn't available it continues looking through the list until it successfully finds a GK.

When the first re-registration message is missed the phone accelerates the rate of sending those message until X consecutive messages are missed at which point . . .

Avaya IP Telephone Use of Gatekeeper List

- IP Telephone looks at DHCP list and then the RAS list received from Communication Manager software to look for Alternate Gatekeeper addresses
- IP Telephone cannot register with LSP until an H.248 MG registers
- IP Endpoints now know about S8300(LSP) though DHCP or RAS process
- Communication Manager provides LSP addresses in RCF based on IP Phone Network Region.
- Need to administer each LSP that needs to register with S8700 (“change lsp” command)
- A network-region can have up to 6 LSPs.

46XX GK Search : What triggers a phone to search for another GK?



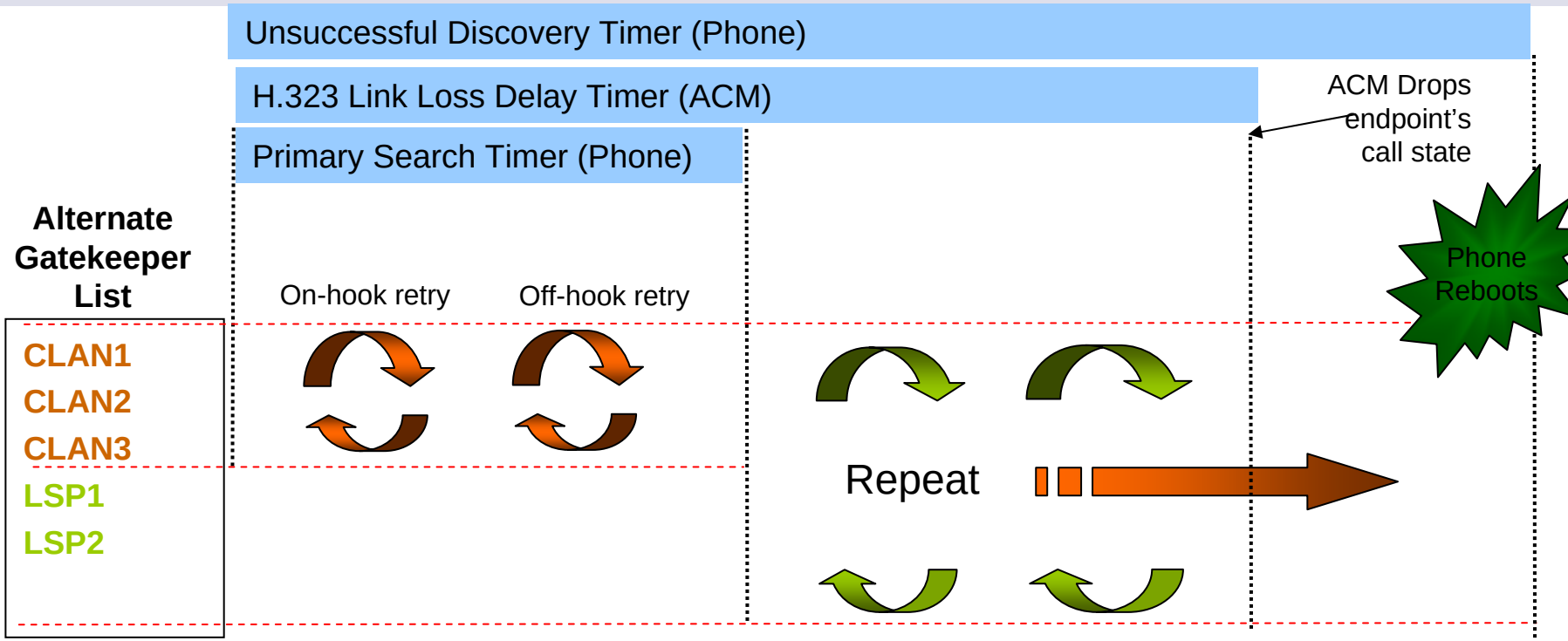
Default Settings:

Idle Traffic Interval -- 20s

Keep Alive Interval – 5s

Keep Alive Count – 5

46XX GK Search

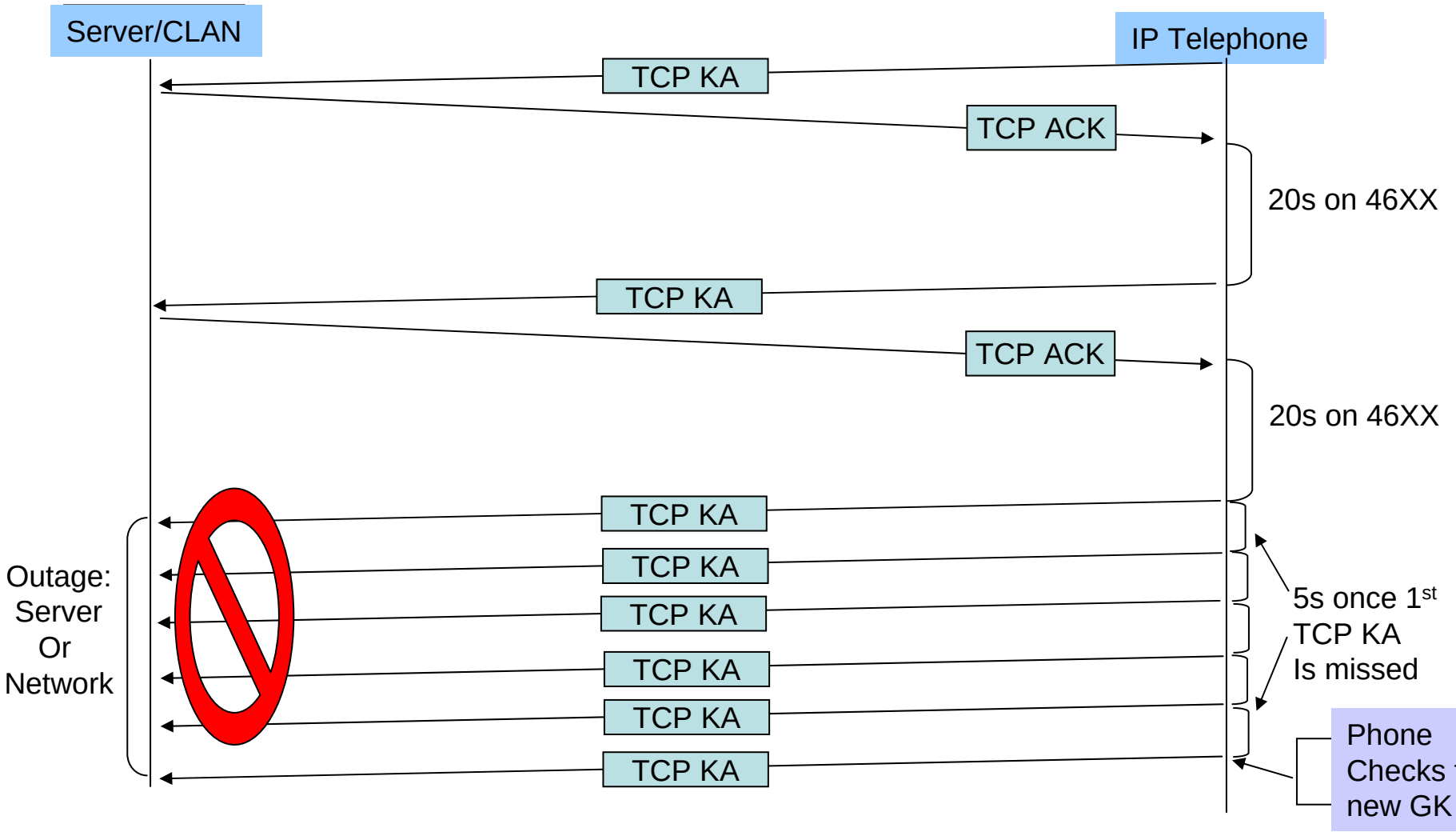


Note : In this example, the Active Gatekeeper (Server) has 3 CLANs And 2 LSPs

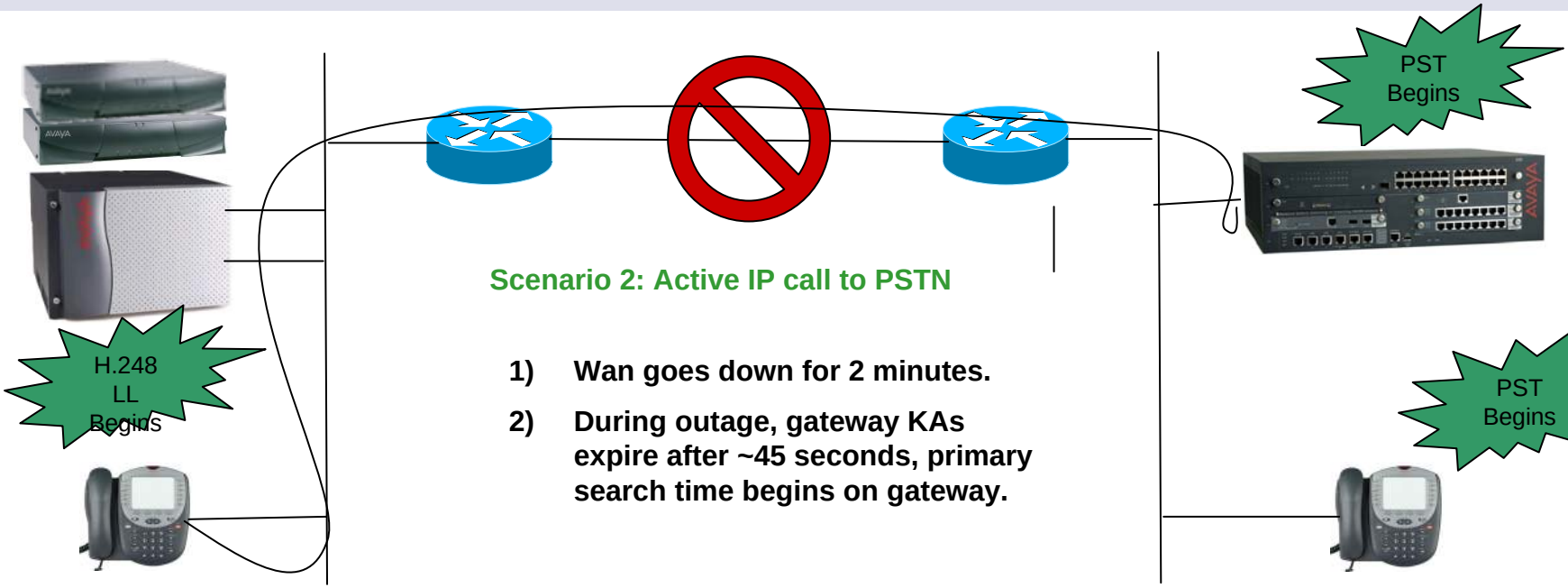
Timer parameters can be found in "system-parameters ip-options" and "ip-network-region" forms. LSP list is found in IP-network-region

During signaling channel loss active calls are preserved. Endpoint attempts to re-register with the original servicing gatekeeper during call duration

46XX GK Search



Timers in Operation (Link Recovery)



Scenario 2: Active IP call to PSTN

1) **Wan goes down for 2 minutes.**
2) **During outage, gateway KAs expire after ~45 seconds, primary search time begins on gateway.**
3) **H.248 link loss timer begins shortly after gateway primary search timer (seconds).**
4) **IP Phone- KAs also expire after ~45 seconds, and H.323 link loss timer also begins shortly after gateway primary search timer (seconds).**
5) **Because WAN recovery occurred prior to the timer expiring phone call is re-established "in progress"**

H.323 Link Loss = 6 min

H.248 Link Loss = 6 min

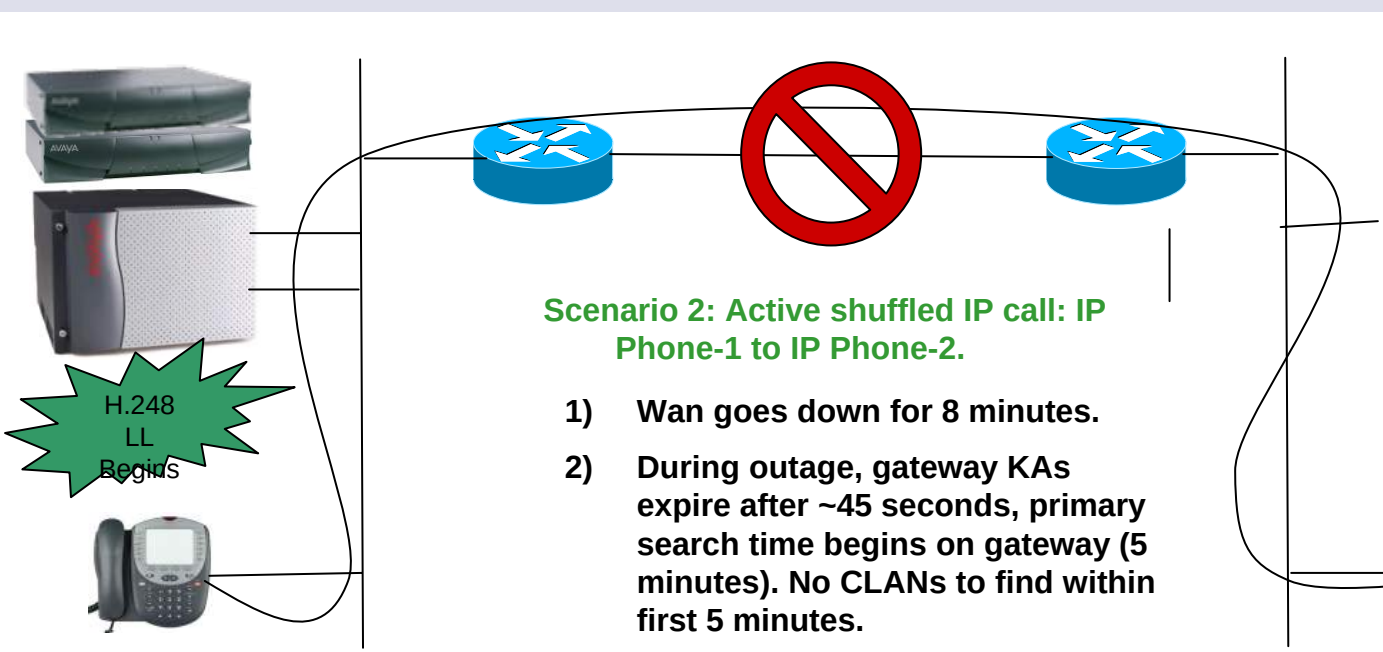
H.323 PST = 5.5 min

H.248 PST = 5 min

Gateway h.248 App. Keep Alive = every 14 seconds, 45 second retry interval.

IP Phone TCP KA = every 20 seconds, 5 retries, 5 seconds each.

Timers in Operation (Timers Expire)



Scenario 2: Active shuffled IP call: IP Phone-1 to IP Phone-2.

1) **Wan goes down for 8 minutes.**
2) **During outage, gateway KAs expire after ~45 seconds, primary search time begins on gateway (5 minutes). No CLANs to find within first 5 minutes.**
3) **H.248 link loss timer begins shortly after gateway primary search timer (seconds).**
4) **IP Phone- KAs also expire after ~45 seconds, and H.323 link loss timer also begins shortly after gateway primary search timer (seconds).**

H.323 Link Loss = 6 min

H.248 Link Loss = 6 min

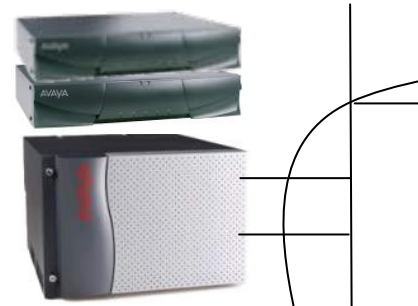
H.323 PST = 5.5 min

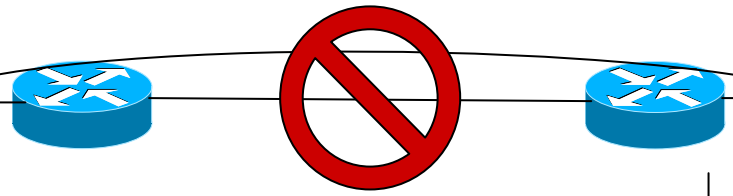
H.248 PST = 5 min

Gateway h.248 App. Keep Alive = every 14 seconds, 45 second retry interval.

IP Phone TCP KA = every 20 seconds, 5 retries, 5 seconds each.

Timers in Operation (Timers Expire)





GW Registers w S8300

Phone Registers w S8300

H.323 Link Loss = 6 min

H.248 Link Loss = 6 min

H.323 PST = 5.5 min

H.248 PST = 5 min

Scenario 2: Active shuffled IP call: IP Phone-1 to IP Phone-2.

5) **Gateway primary search timer expires after 5 min, gateway moves beyond TP, gateway registers to LSP, and MGC resets.**

6) **Phone Primary Search Timer expires after 5.5 minutes (transitions to LSP)**

7) **H.248 an H.323 Link Loss timers expire. Resources are liberated on primary call server. Calls can no longer be re-instated.**

8) **WAN returns after 8 total minutes. Phones now registered to discrete call processors.**

Future: Connection Preserving Transition

Gateway H.248 App. Keep Alive = every 14 seconds, 45 second retry interval.

IP Phone TCP KA = every 20 seconds, 5 retries, 5 seconds each.

QOS Across the Enterprise

Voice Application in a Data Network

What happens when you put voice in your data network?

- **Data Communication is Bursty in Nature**
- **Packet Networks are Asynchronous**
- **Voice is a Real-Time Application**
- **Voice Transmission is Synchronous**

What is the solution?

- **A Voice ready Network <u>needs</u> QoS**
- **Long Term Solution should be Policy Based**

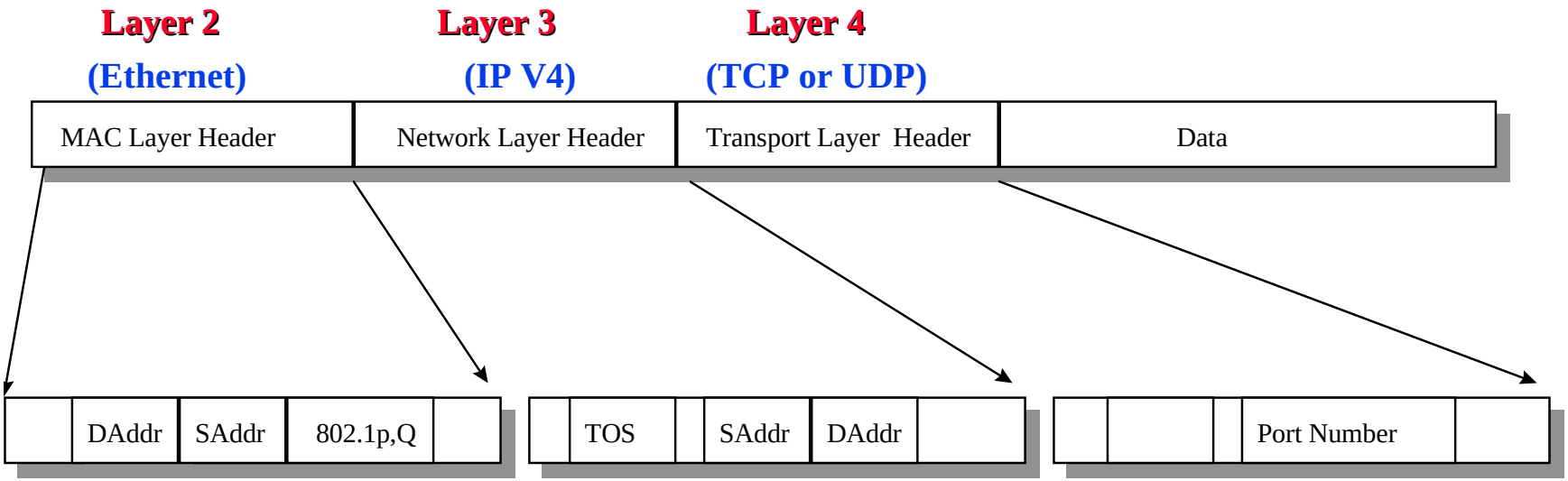
Class of Service – Prioritization (tagging)

Quality of Service – Processes in place to assure the prioritized packet get to destination

Avaya QoS solution:

- **Layer 2 : 802.1p/Q – VLAN and priority inside the VLAN**
- **Layer 3 : DiffServ (TOS byte), RSVP – WAN queuing, Bandwidth Reservation**
- **Layer 4 : UDP Port Range – No suggested range**
- **100% Standards Based**

Standards Based Class of Service



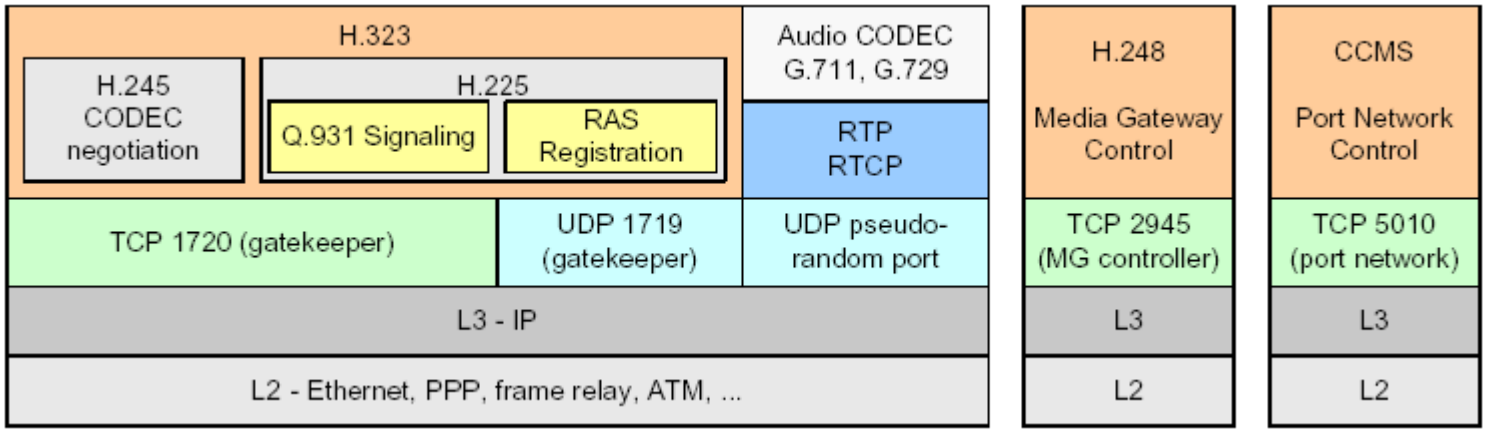
802.1p specifies priority desired

TOS field specifies service level desired

Saddr/Daddr or Saddr/Daddr/Port # identifies RSVP flow

Port Number identifies application/session

Protocols and Ports



Registration (H.225 RAS) = UDP 1719

Signaling (H.225 Q.921) = TCP 1720

Voice (RTP) = UDP 2048-65535 (configurable)

Media Gateways (H.248) = TCP 2945

Port networks (“classic” media gateways) = TCP 5010

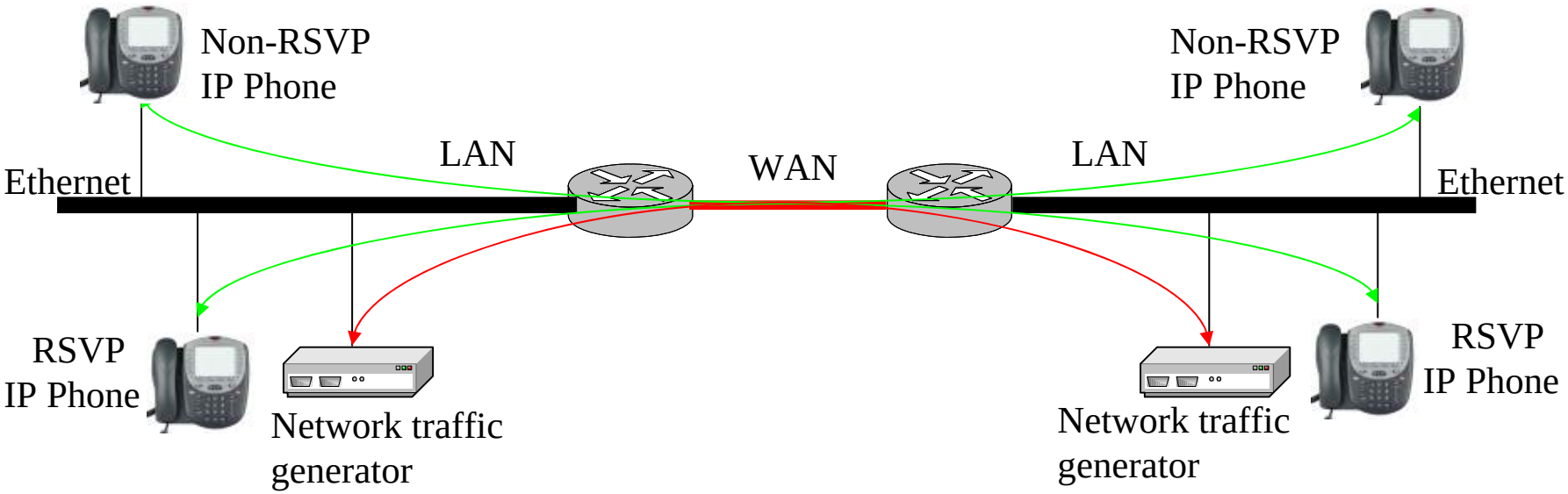
QoS Requirements

- **Delay (one way between endpoints):**
  - ITU spec is 150ms or less
  - Avaya recommends 80ms or less for “business quality audio”
  - Delay over 150ms could be acceptable depending on customer expectations, codec, etc.
  - Delay over 250ms causes “talk over” problems
- **Jitter (variation in delay):**
  - Less than 20ms recommended
  - Defaults can handle up to 30ms (dependent on sampling rate)
- **Packet loss:**
  - Less than 1% recommended

RSVP - Resource reSerVation Protocol

- RSVP is a QoS signaling protocol
- RSVP/Integrated-Services provides protection for the voice bearer channel in a loaded or congested network.
- IP Phones/Gateways request the network routers to reserve bandwidth.
- Routers act upon the request to allocate bandwidth according to QoS request.
- When bandwidth is reserved, the call is protected against other network traffic.
- This ensures good voice quality for the users.

RSVP in action



1) RSVP enabled phone call is established.

2) RSVP disabled phone call is established.

3) When the network is loaded with emulated voice traffic:

RSVP enabled bearer channel is protected exhibiting good voice quality

RSVP disabled bearer channel is not protected exhibiting bad voice quality

When to enable RSVP

- If the customer wants a scenario where N calls get guaranteed service and the N+1th call competes with everything else, then RSVP is the best solution.

- But if the customers want a scenario where N calls get guaranteed service and the N+1th call is not permitted to go through, then Call Admission Control schemes need to be used.

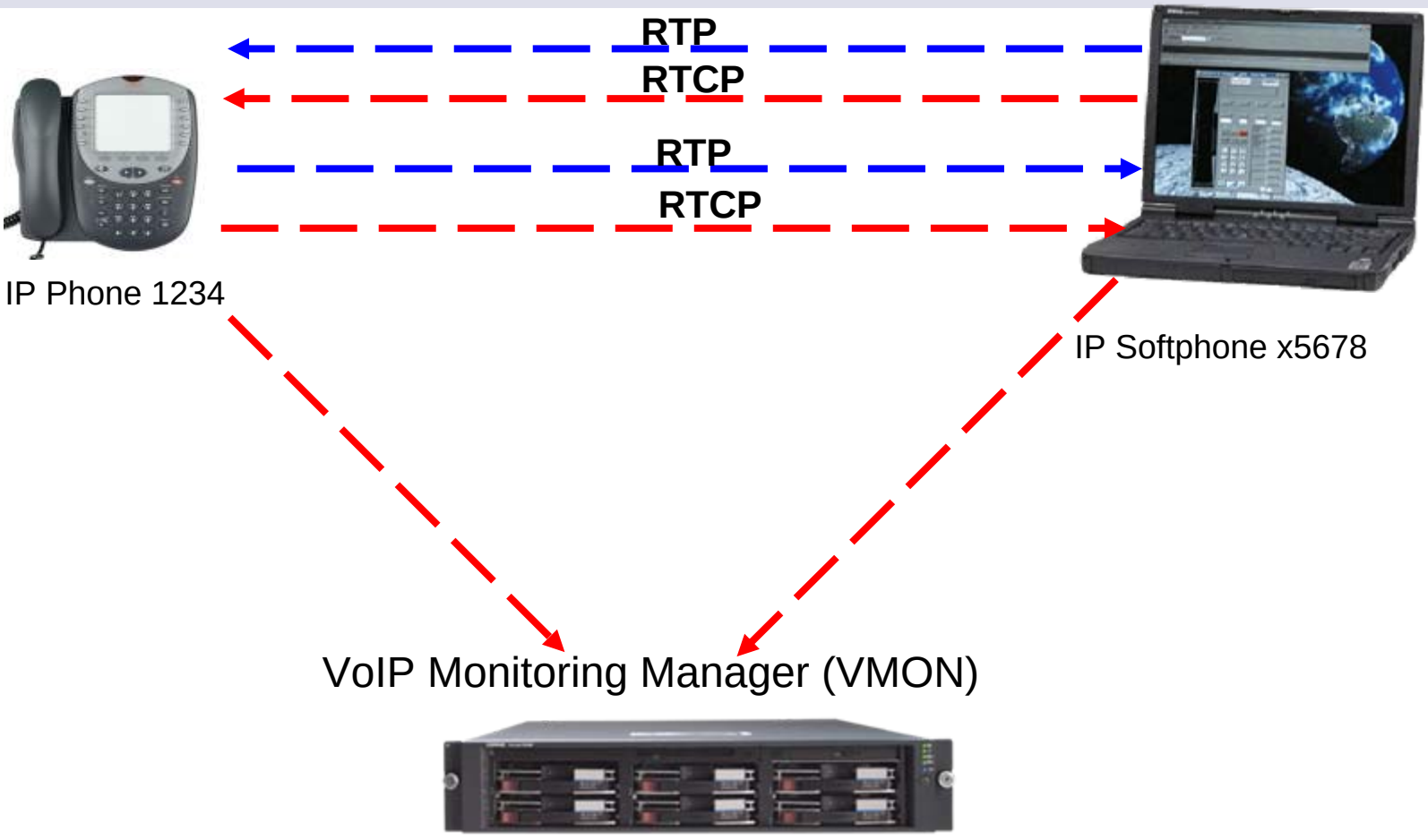
So What If I'm Experiencing Poor Quality Voice

- Factors that need to be examined
  - Network Metrics (Packet Loss, Jitter, Latency)
  - Trunk connectivity (Digital, Analog)
  - DSP resources (Medpro, Gateway)
  - End User Device (Headset, Terminals)
  - Environmental
- Psychology may be a factor
  - People are more alert after a change
  - Feature Issues

Symptoms & Possible Causes

Symptom	Possible Cause
Echo	Trunks, Latency
Tininess	Packet Loss, Jitter
Static	Packet Loss, Stations
Muffled, Garbled	Stations
Volume Levels	Trunks, Environment, Stations
Clipping, “Breaking up”	Packet loss, Silence Suppression

VoIP Monitoring Manager (VMON)

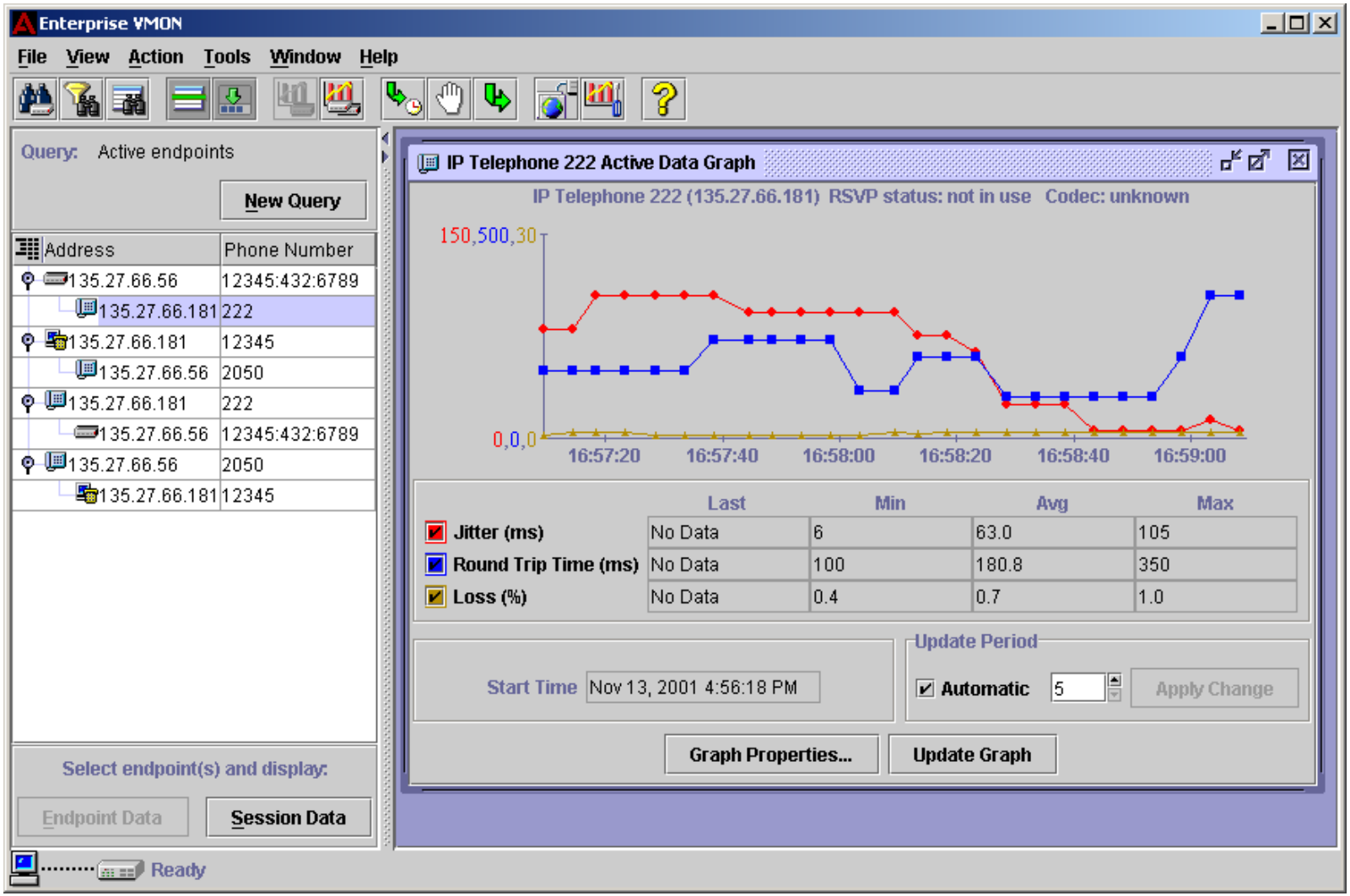


RTCP (Real Time Control Protocol) – RFC 1889

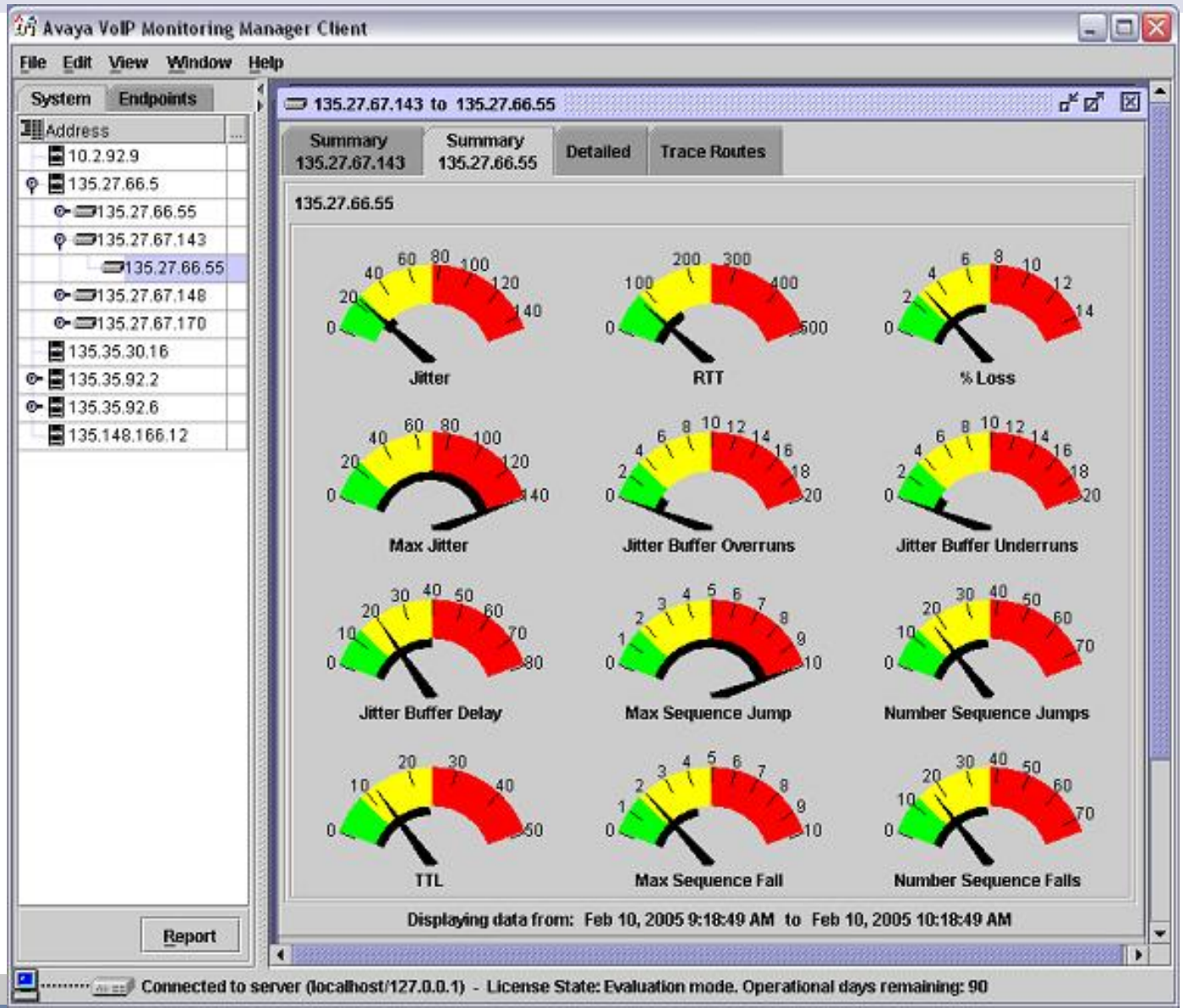
QoS Monitoring with VMON

- What it does?
  - Record call statistics (delay, jitter and packet loss) on some or all calls (configurable by network region)
  - Real-time view or Historical (up to 30 days at this time). Search by extension number, time range or IP address
  - Configurable SNMP traps for different combinations or jitter, delay and/or packet loss thresholds
- What do I get out of it?
  - Baseline: What did things look like before or after the change?
  - Troubleshoot: Comparing different groups of endpoints.
  - Proactive Monitoring: Be alerted if service falls below a certain level.

VoIP Monitoring Manager (VMON)

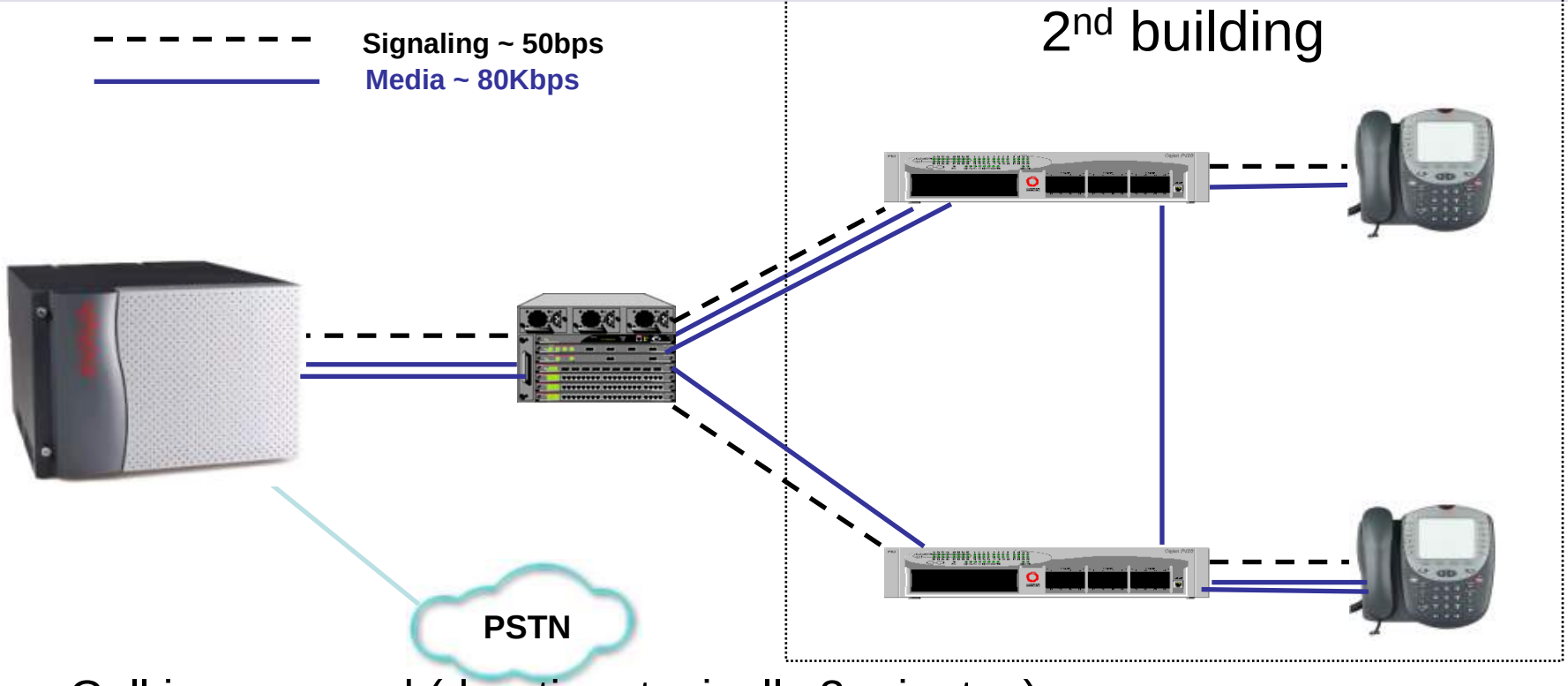


VoIP Monitoring Manager (VMON)



Bandwidth Considerations

Shuffling



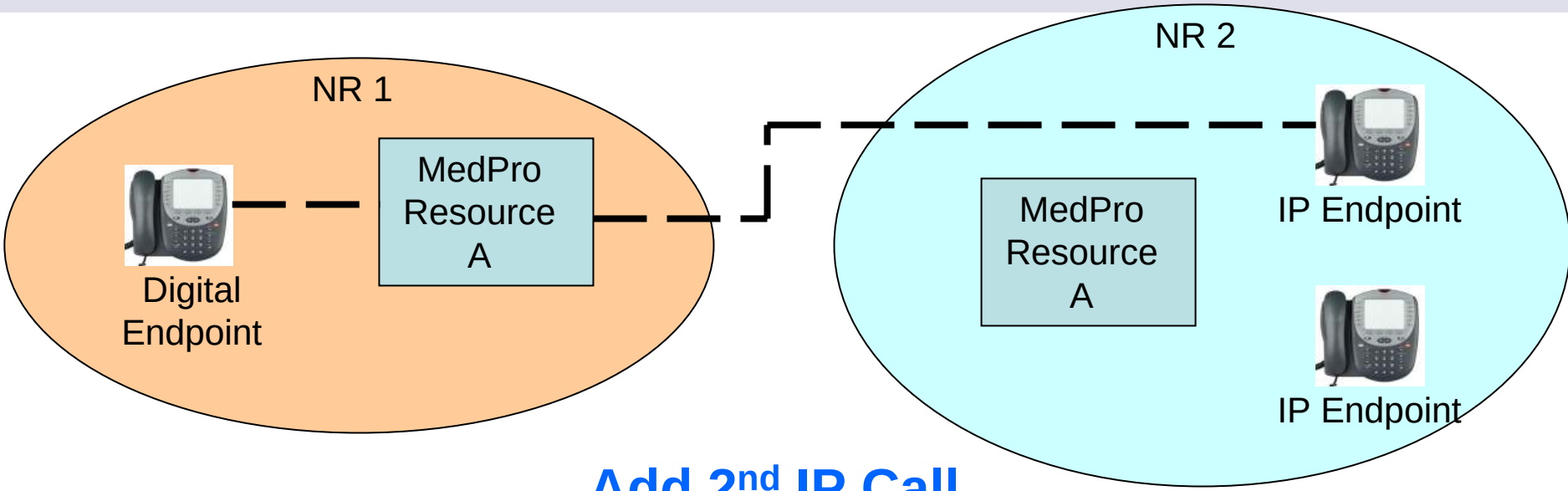
Call is answered (duration: typically 3 minutes):

- Avaya Call Processor sees that both endpoints are IP, and asks A to ring B, and vice versa.
- When A answers, the ACP tells the phones to send voice packets to each other, but keeps signaling

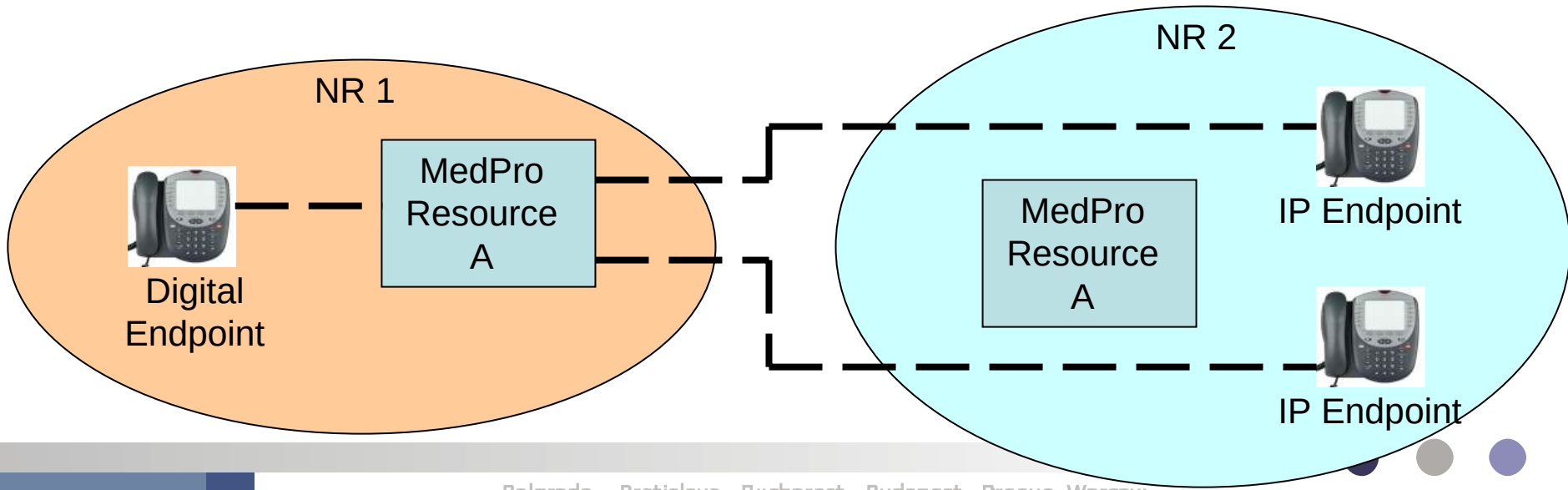
Call is in Conference (duration: typically 3 minutes):

Avaya Call Processor knows that now it needs to mix the calls, so it redirects the media piece off the phones and into the Media Processor (MedPro)

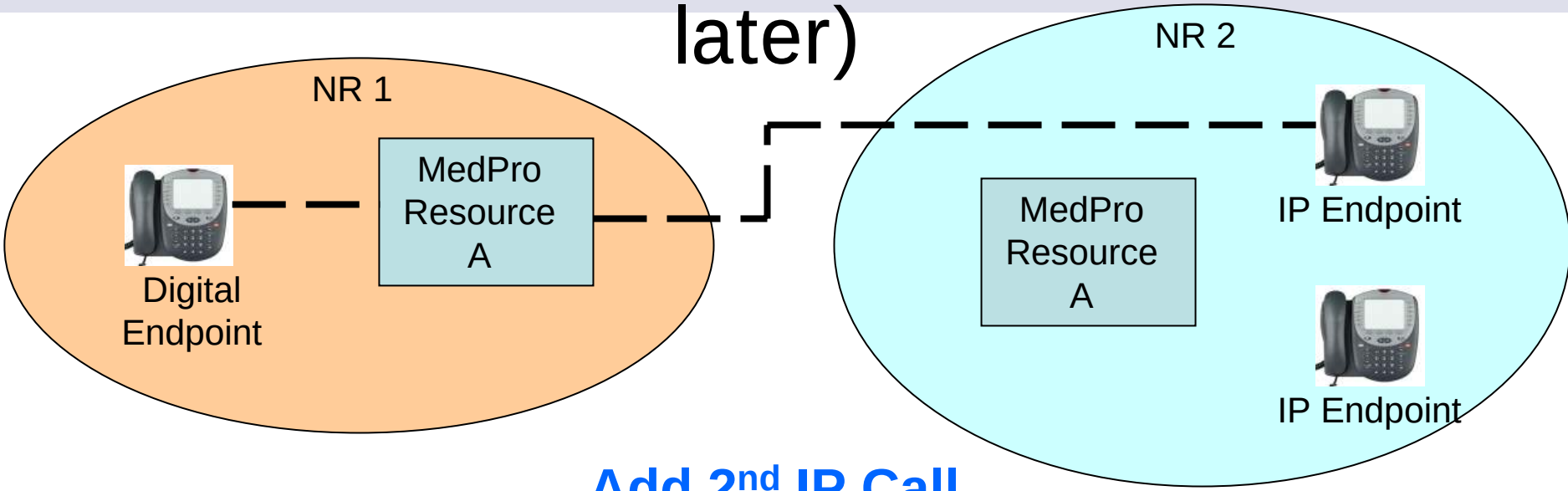
Conferencing Scenario (Pre ACM 3.0)



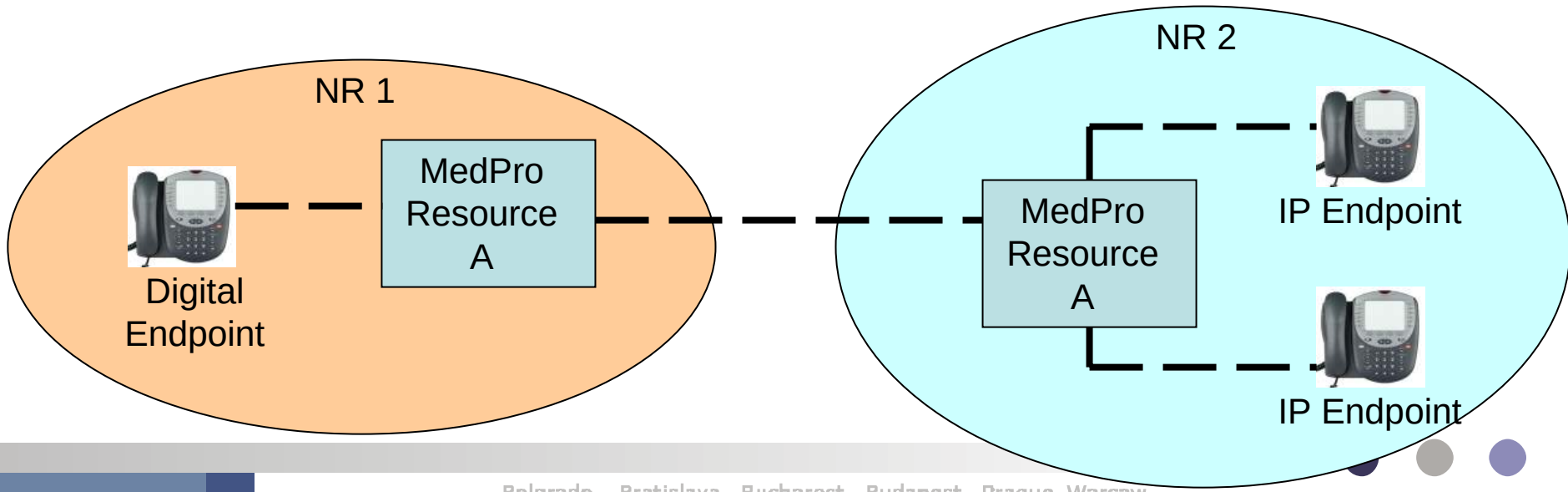
Add 2nd IP Call



Conferencing Scenario (ACM 3.0 and later)



Add 2nd IP Call



Bandwidth Considerations

- Bandwidth impact on a LAN/WAN depends on
  - CODEC used
    - G.711 which produces 64Kbps voice samples
    - G.729 which produces 8 Kbps voice samples
    - G.723.1 which produces 6.3 and 5.3 Kbps voice samples
  - Frame size used
    - G.711 uses 10ms frames (80 bytes)
    - G.729 uses 10ms frames (10 bytes)
    - G.723 uses 30ms frames
  - Number of Frames per packet
  - Protocol Overhead

Minimize # codec sets
LAN Codec Set
(G.711 20ms samples, modem pass-through)
WAN Codec Set
(G.729 30ms samples, modem relay)

G.711 Analysis

- G.711 uses 64Kbps voice samples
  - 64000bps equals 64 bits per ms
  - 64 bits per ms equals 8 bytes per ms
- A G.711 Frame is 10 ms or 80 bytes
- Protocol overhead

Uncompressed Real Time Protocol (RTP)	12 Bytes
User Datagram Protocol (UDP)	8 Bytes
Internet Protocol (IP)	20 Bytes
Layer 1 and 2 Ethernet	26 Bytes
TOTAL	66 Bytes

Ethernet Header Breakdown

- Ethernet has the following components:

Preamble and 1 byte start of frame delimiter	8 Bytes
Ethernet (Type, MAC SRC, MAC DST)	14 Bytes
802.1Q (priority and VLAN)	4 Bytes

Data Network Impact of Active G.711 IP Call

G.711(64Kbps) Number of 10 ms Frames per Packet	Packet Size	**Audio Payload** **(Codec Frame size*Packet Size)**	Total Packet Size (Audio Payload plus packet overhead)	Total Bandwidth (Kbps) (Total Packet Size*8 =bps/packet size)	Target Delay (msec)
1	10ms	80 Bytes	146 Bytes	116.8	62ms
2	20ms	160 Bytes	226 Bytes	90.4	72ms
3	30ms	240 Bytes	306 Bytes	81.6	82ms
4	40ms	320 Bytes	386 Bytes	77.2	92ms
5	50ms	400 Bytes	466 Bytes	74.5	102ms
6	60ms	480 Bytes	546 Bytes	72.8	112ms

Bandwidth Minimization

- Three approaches to minimize bandwidth
  - Choose a low bit rate audio codec
  - Combine multiple audio frames into one packet
  - Suppress transmission of silence
  - Use header compression
- Lower bit rate codec can degrade quality and increase processing
- Combining multiple audio frames in one packet reduces bandwidth required
- Combining multiple audio frames in one packet increases delay

Bandwidth for Different Size Voice Samples

Sample Size (ms)	G.711	G.729
10	96.0	40.0
20	80.0	24.0
30	74.7	18.7
40	72.0	16.0
50	70.4	14.4
60	69.3	13.3

- Default is 20ms (which is the recommended setting for most situations)
- Smaller samples make it less efficient (more bandwidth consumed)
- Larger samples make it more efficient… BUT at a cost….
  - Increases latency
  - A greater amount of voice is lost if packet loss occurs

Full and Half Duplex Facilities

Full Duplex: Transmit and Receive Simultaneously

(WAN Facilities and Switched Ethernet)

AND

Half Duplex: Can Either Transmit or Receive

(Shared Ethernet)

OR

Bandwidth Impact on Full Duplex Facilities

CODEC TYPE (30ms Packets)	A and B Both Suppress Silence	A Suppresses Silence and B Does Not	Neither End Suppresses Silence
G.711	A Talking to B 80 Kbps 0 Kbps	A Talking to B 80 Kbps 80 Kbps	A Talking to B 80 Kbps 80 Kbps
	B Talking to A 0 Kbps 80 Kbps	B Talking to A 0 Kbps 80 Kbps	B Talking to A 80 Kbps 80 Kbps
G.729	A Talking to B 24 Kbps 0 Kbps	A Talking to B 24 Kbps 24 Kbps	A Talking to B 24 Kbps 24 Kbps
	B Talking to A 0 Kbps 24 Kbps	B Talking to A 0 Kbps 24 Kbps	B Talking to A 24 Kbps 24 Kbps

*** SS and VAD conserve bandwidth at the price of voice clipping potential

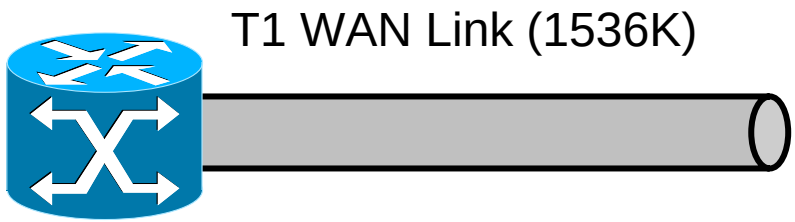
Compression of RTP header

Codec	Payload bytes/packet	Packets /sec	Avg WAN BW consumption (kbps)		% reduction
			w/o compression	w/ compression	
G.711 (64 kbps)	160	50	84	68.5	~18 %
G.729A (8 kbps)	20	50	27.5	13	~53%
G.723.1 (5.3 kbps)	20	33	18	9	~50%
G.723.1 (6.3 kbps)	24	33	19	10	~47%

Router Considerations

Router Throughput

What Factor Most Greatly Determines Router Performance?



**** Full duplex loading is uncommon for data environments, but 'typical' for voice.***

Typical Data Application

Packet size - 60 to 1500 bytes
Average - ~ 300 bytes

Full T1 = 1536K*2 / (300 * 8)
= **~ 1,280 PPS**

Typical VoIP Application

Packet size - 86 bytes

Full T1 = 1536K*2 / (86 * 8) =
~ 4,465 PPS

Make Sure Your Routers Can Handle A Greater Number of PPS

cRTP, MLPPP Significantly More CPU cyles

WANs that Contain ATM

- For a G.729 Sample use 30ms samples instead of 20ms (more common)
  - Packet Rate reduced from 50 to 33.33 PPS
  - Still fits in 2 ATM Cells
- Effective ATM bandwidth
  - 2 cells * 33.33 PPS = 2 * 53 * 8 *33.333 =

    28.26K / Call

    33.33 PPS / Call
- *Advantages*
  - *Reduces Router CPU Load*
  - *Close to FR per call bandwidth*

Little Known Fact : Many SP Networks (including MPLS) Still Utilize ATM

Call Admission Control

Overview of Call Admission Control

- Provides ability to block Voice over IP (VoIP) calls that go between IP Network Regions
  - IP Network Regions generally interconnected by WAN links
  - WAN links are lower bandwidth facilities
  - IP Network Region pairs can be directly connected or indirectly connected via intervening IP Network Regions
- Blocking calls when bandwidth is full helps ensure Quality of Service (QOS) for existing VoIP calls
- Applies only to bearer traffic, and not to data or signaling traffic from CM or other customer traffic
- Does not apply within an IP Network Region
  - Unlimited bandwidth is assumed

Offer Considerations

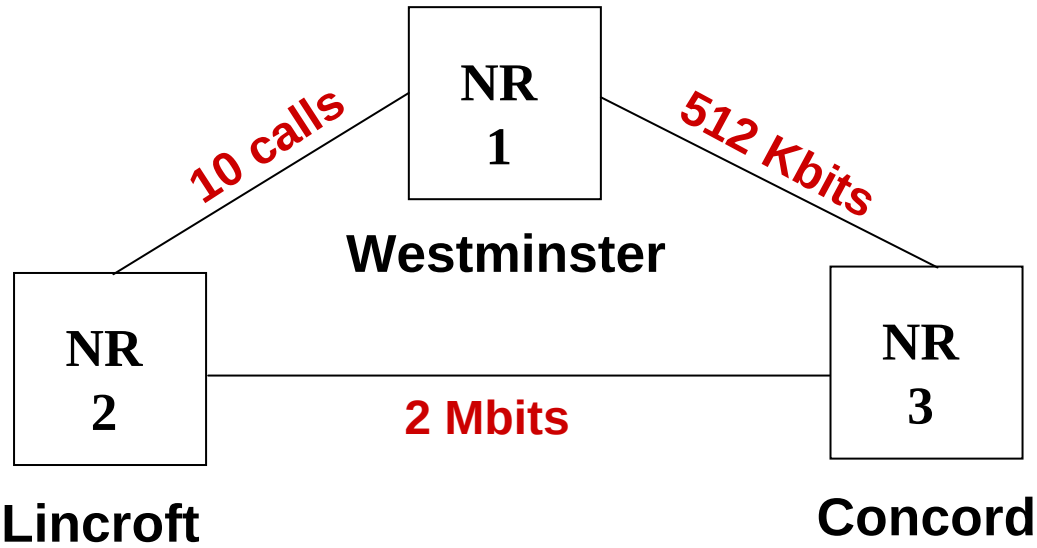
- Available in ACM 2.0 and later
- Supported in Linux platforms (S8300, S8500, S8700)
  - All gateways
- One point of administration for system
  - No need to configure individual parameters across routers
- Not a substitute for other QOS (I.e. Diffserve, 802.1p/Q)

Call Admission Control Functionality

- Administer optional bandwidth limits between IP Network Regions
- Applies to all VoIP calls between the IP Network Regions for:
  - Stations
  - Trunks
  - Port Networks
  - Media Gateways
- CM software keeps track of bandwidth used for IP bearer traffic between IP Network Regions (direct or indirectly connected)
  - Direct use bandwidth on a single link
  - Indirect use bandwidth on multiple links
- Attempts to make VoIP connections that would cause bandwidth limits to be exceeded will be blocked
  - ACM 3.0 is targeted to include Alternate Routing

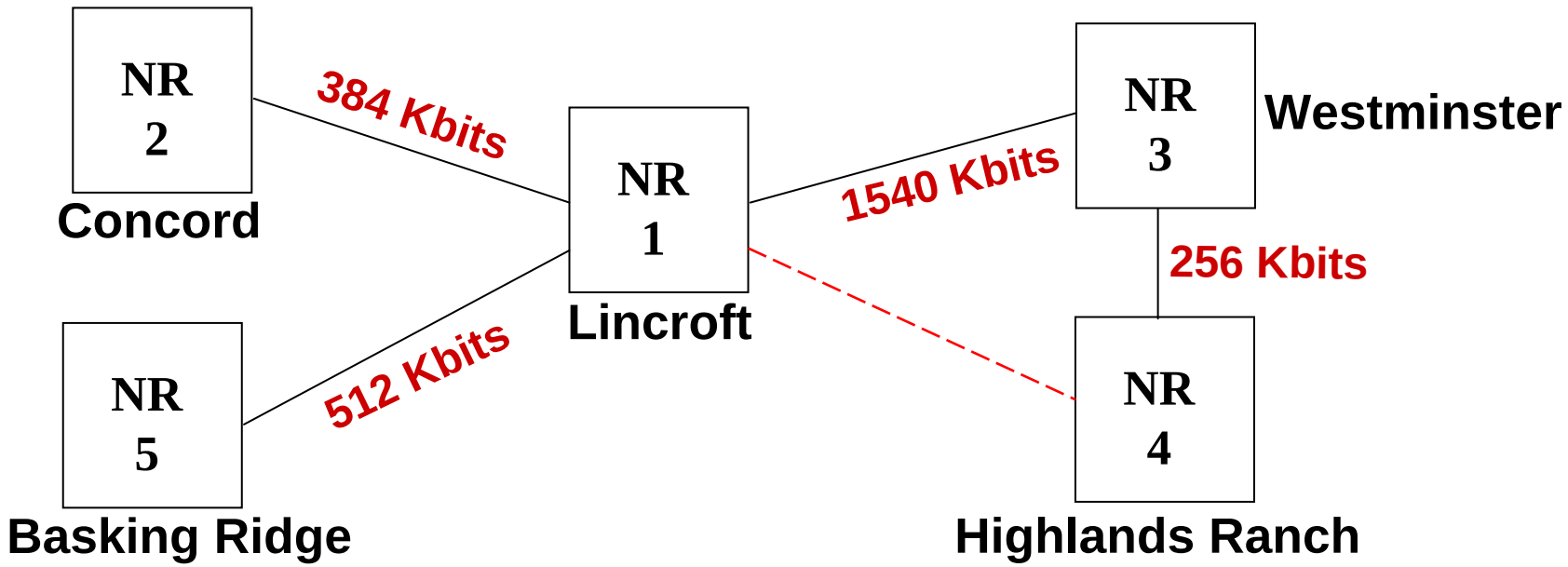
IP Network Regions Configurations - Directly Connected

- IP Network Regions (NR) 1 and 2 and 3 are all directly connected
  - Administer bandwidth limits between NR1 and NR2, NR1 and NR3, and NR2 and NR3

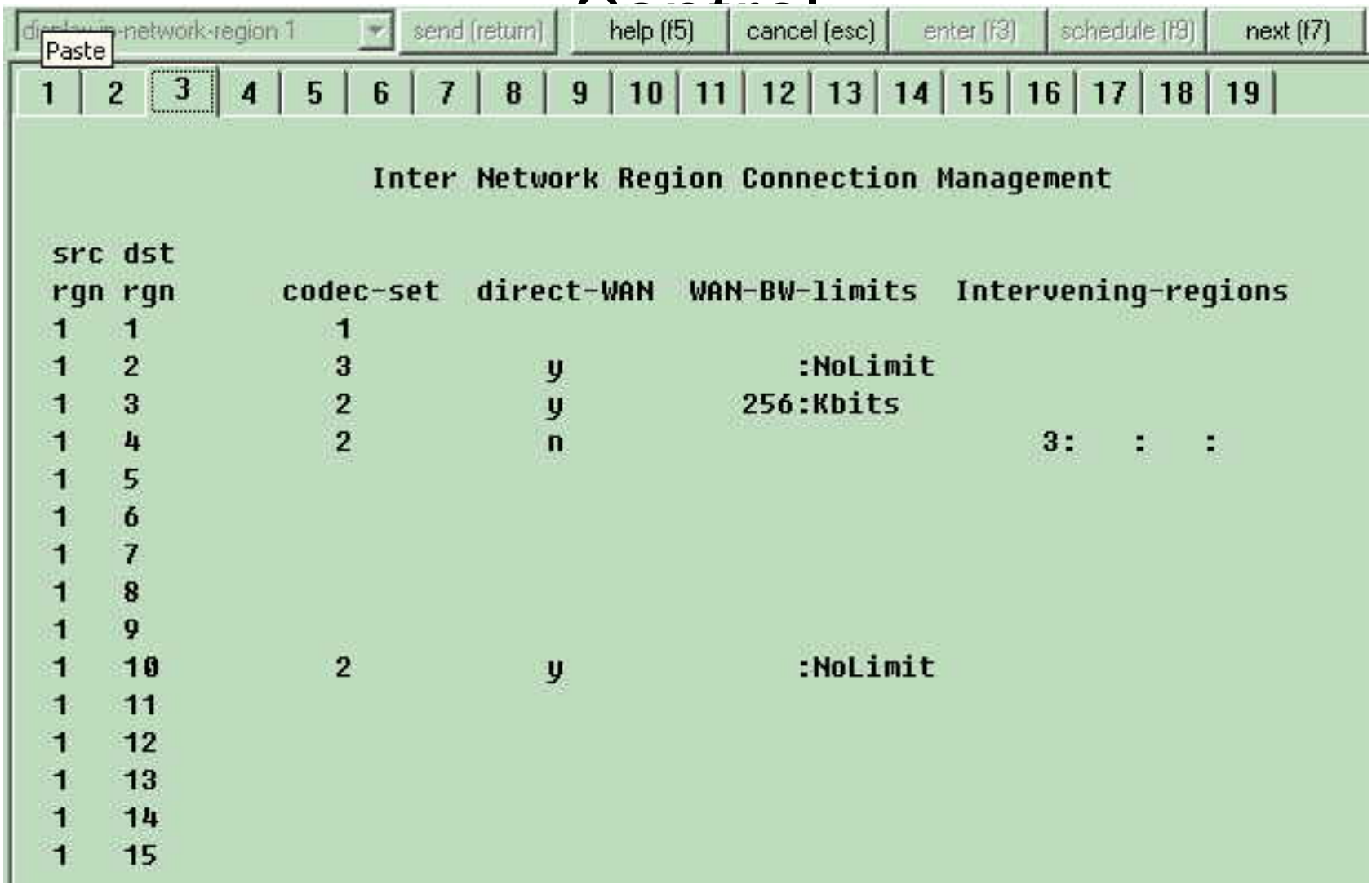


IP Network Regions Configurations - Indirectly Connected

- Administer direct connectivity between NR1 and NR2, NR1 and NR5, NR1 and NR3, and NR3 and NR4
- Administer intervening regions for all others
  - For example, Basking Ridge connects to Highlands Ranch via the link to Lincroft, then via the link to Westminster, and then via the link to Highlands Ranch (e.g. 5 to 1 to 3 to 4)
  - Only 1 path can be administered



Administration of Call Admission Control



Bandwidth Limits

- Bandwidth Limits can be administered in units of:
  - Number of connections
  - Kbits/second
  - Mbits/second
  - No limits
- Some networks are better suited for limits based on number of connections instead of bandwidth
  - Only one codec used between regions - use connections
  - Multiple codecs used between regions - use bandwidth
  - Silence suppression – use connections

Bandwidth Usage

- Bandwidth Usage per call is a function of:
  - Codec set (e.g. G.711, G.729, etc.)
  - Packet size
  - Assumes 7 byte L2 WAN header

Bandwidth Usage (kbits/sec)

Packet Size	10ms	20ms	30ms	40ms	50ms	60ms
G.711	102	83	77	74	72	71
G.729	46	27	21	18	16	15
G.723-6.3	NA	NA	19	NA	NA	13
G.723-5.3	NA	NA	18	NA	NA	12

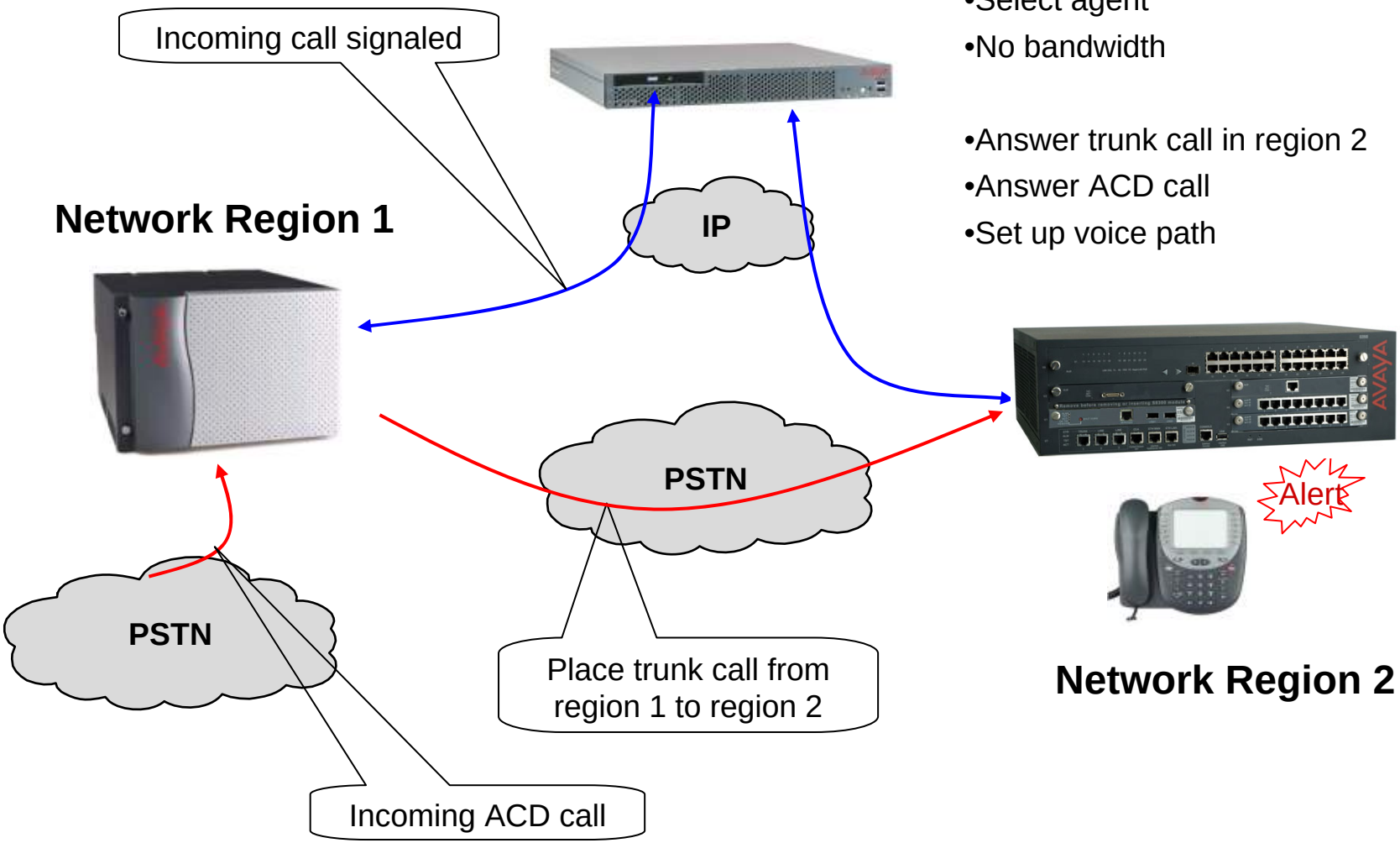
Additional Bandwidth Considerations

- In general, bandwidth is used in both directions – except for the following (one direction only):
  - Announcements
  - Music on Hold
  - Firmware download to port boards – uses bearer channel from CLAN board to port board
- No adjustment in bandwidth made for FAX calls
  - Uses bandwidth as determined when initialing setting up the call
- No adjustment for call on hold
  - Bandwidth is reserved
- No adjustment made for silence suppression

When Calls are Blocked via CAC-BL

- Calls blocked by CAC-BL (bandwidth limit) can be routed to an alternate destination via:
  - Hunting
  - Call coverage paths
  - Another trunk group as administered in routing patterns
- If blocked call is not routable, caller will get reorder tone when possible
- No automatic routing of blocked calls via PSTN facilities to the desired destination in CM2.0
  - Alternate routing targeted for Avaya Communication Manager 3.0 release

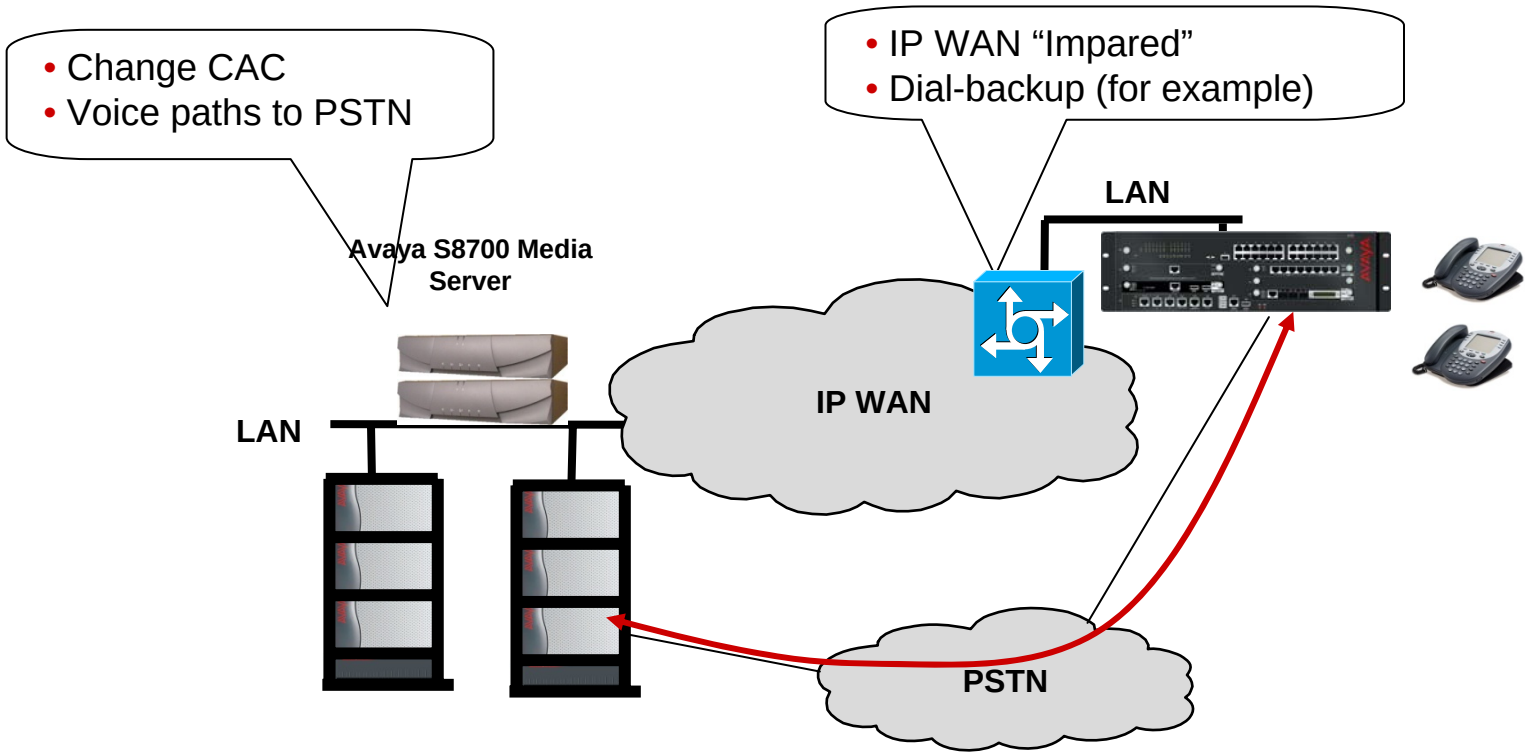
Alternate Routing Scenario



- Select agent
- No bandwidth

- Answer trunk call in region 2
- Answer ACD call
- Set up voice path

Dynamic CAC


• Change CAC
• Voice paths to PSTN
• IP WAN “Impared”
• Dial-backup (for example)
Avaya S8700 Media Server
LAN
LAN
IP WAN
PSTN

Media Encryption

What is Media Encryption?

- Encryption of the VoIP RTP bearer
- Uses H.235 extensions to H.323
- Encryption capabilities negotiated between H.323 Endpoints and H.323 Gatekeepers
- Avaya was the first to offer such security to VoIP customers (with AEA Media Encryption)
- CM2.0+ now includes encryption using the “Advanced Encryption Standard” (AES)

Why AES Media Encryption?

- AEA Media Encryption:
  - Based upon Avaya patented encryption algorithm
- AES Media Encryption:
  - AES is currently specified by the IETF as the required encryption algorithm for a new internet standard for secure RTP - SRTP.
  - SRTP employs AES encryption to encrypt RTP messages.
  - Will position Avaya products so that they can quickly transition to SRTP
- Some vendors proclaim to be SRTP compliant but in reality they only offer it between their most expensive phones – and not between gateways and phones.

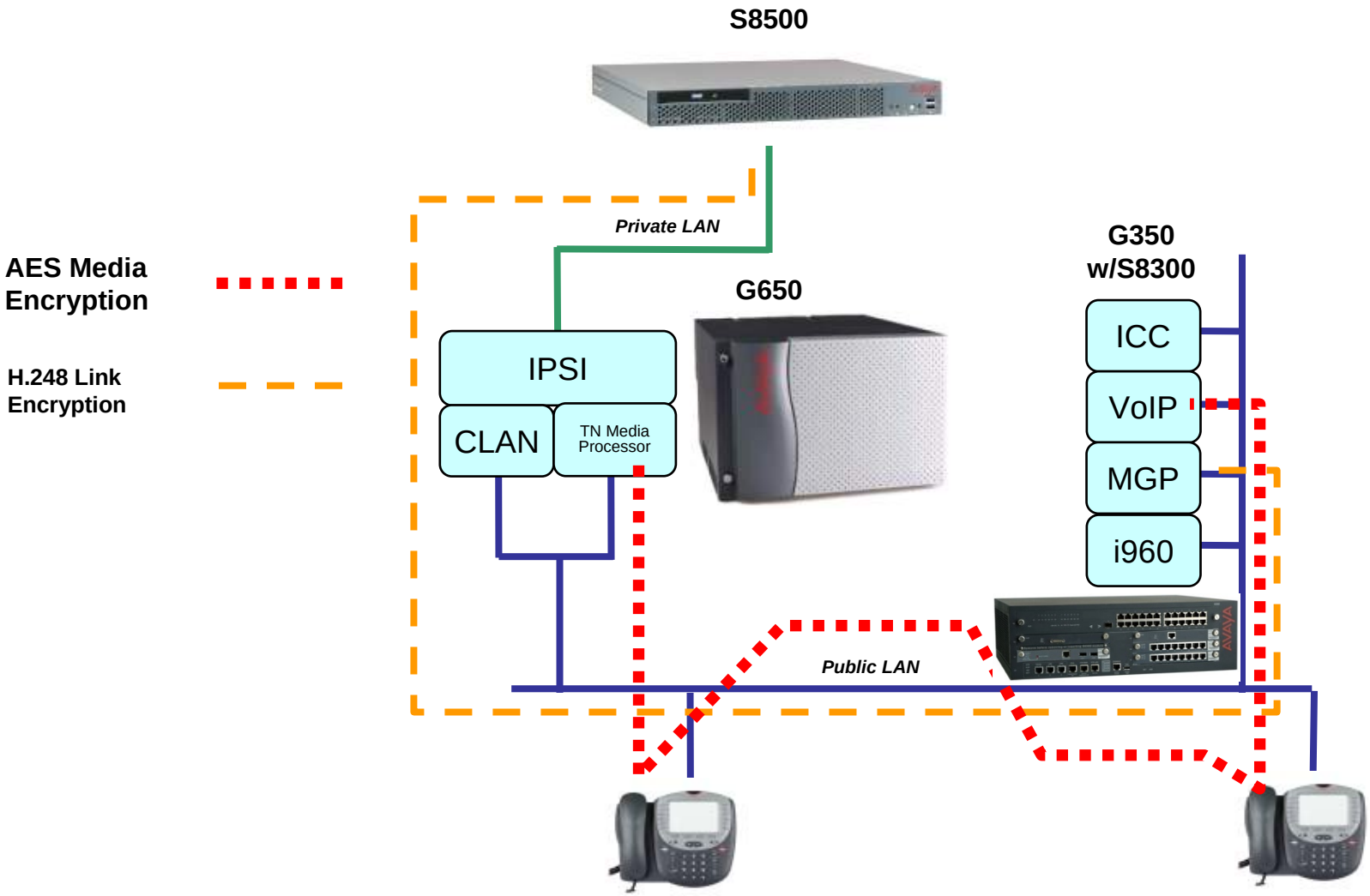
How Media Encryption Works

- During establishment of the call signalling channel, H.323 Endpoint support for media encryption is specified in H.245 elements
- During call setup, H.323 Gatekeeper determines media encryption requirements for call (c.f. codec determination)
- If H.323 Gatekeeper determines that media encryption is to be applied to a call, it will specify via H.245/H.235:
  - What encryption algorithm to use
  - What key material to use

How Media Encryption Works

- The key material to use is encrypted prior to sending to the H.323 Endpoint
  - The encryption of the key material is done using 3DES
  - The station security code/PIN is used as the key for the 3DES encryption
- Encryption of the VoIP RTP payload is between:
  - IP Endpoint – Gateway
  - IP Endpoint – IP Endpoint
  - Gateway - Gateway
- Media Encryption has NO effect on Voice Quality and NO noticeable effect on delay

How Media Encryption Works



Supported Platforms

- Server CSI, Server SI, Server R
- S8100, S8300, S8500, S8500, S8700
- TN2302AP - H/V3 (“TN Media Processor”) and H/V11 (“Cruiser”)
  - Due to the algorithmic complexity of AES algorithm, a 25% reduction in channel capacity on MM760 and TN Media Processor/Cruiser boards will result:

	Number of Simultaneous Calls		
	Encryption Algorithm		
Codec	None	AEA	AES
G.711	64	64	48
G.729/723	32	32	24

- **(Once again): Media Encryption has NO effect on Voice Quality and NO noticeable effect on delay**

Supported Platforms

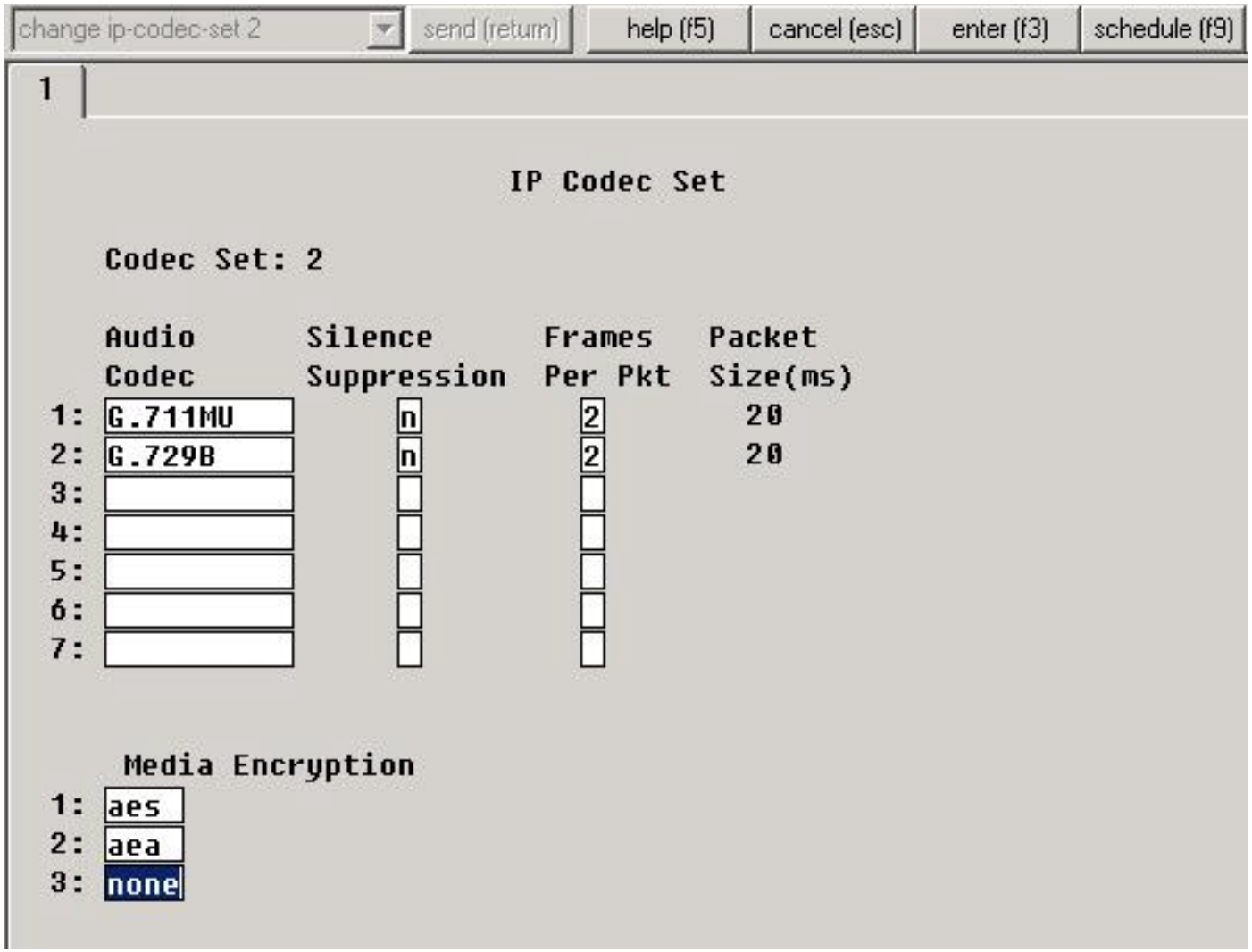
- IP Telephones (4602/4606/12/20/24/30) – AEA Only
- IP Telephones (4610/20/30) - AES and AEA supported.
- IP SoftPhone/SoftConsole/Agent

- G350/G700
  - H.248 Link Encryption must be enabled (for media session key exchange)

Administration

- The Media Encryption feature is controlled by RFA
  - The ‘Media Encryption Over IP’ customer-option must be enabled for any Media Encryption features to work
- H.323 signalling-group Administration
  - Media Encryption must be enabled
  - A Passphrase must be specified
- ip-codec-set Administration
  - 3 options: aes, aea, none

Ip-codec-set Administration



H.323 signaling-group Administration

- Enable (y) ‘Media Encryption’ (default is disabled (n))

- Specify a ‘Passphrase’
  - 8-30 characters. Can include ‘ !&*?;’^(),.:-’. At least 1 alphabetic and 1 numeric character
  - Must use the same ‘Passphrase’ on the Near-end and Far-end signalling-group forms
  - The ‘Passphrase’ is used to 3DES encrypt the key material prior to transmission to the other end (c.f. station security code/PIN)

- Media Encryption selection is still controlled by the administration of the near-end and far-end ip-codec-sets for the specific network regions

Q&A

SIP Enablement Services Basic Administration

Agenda

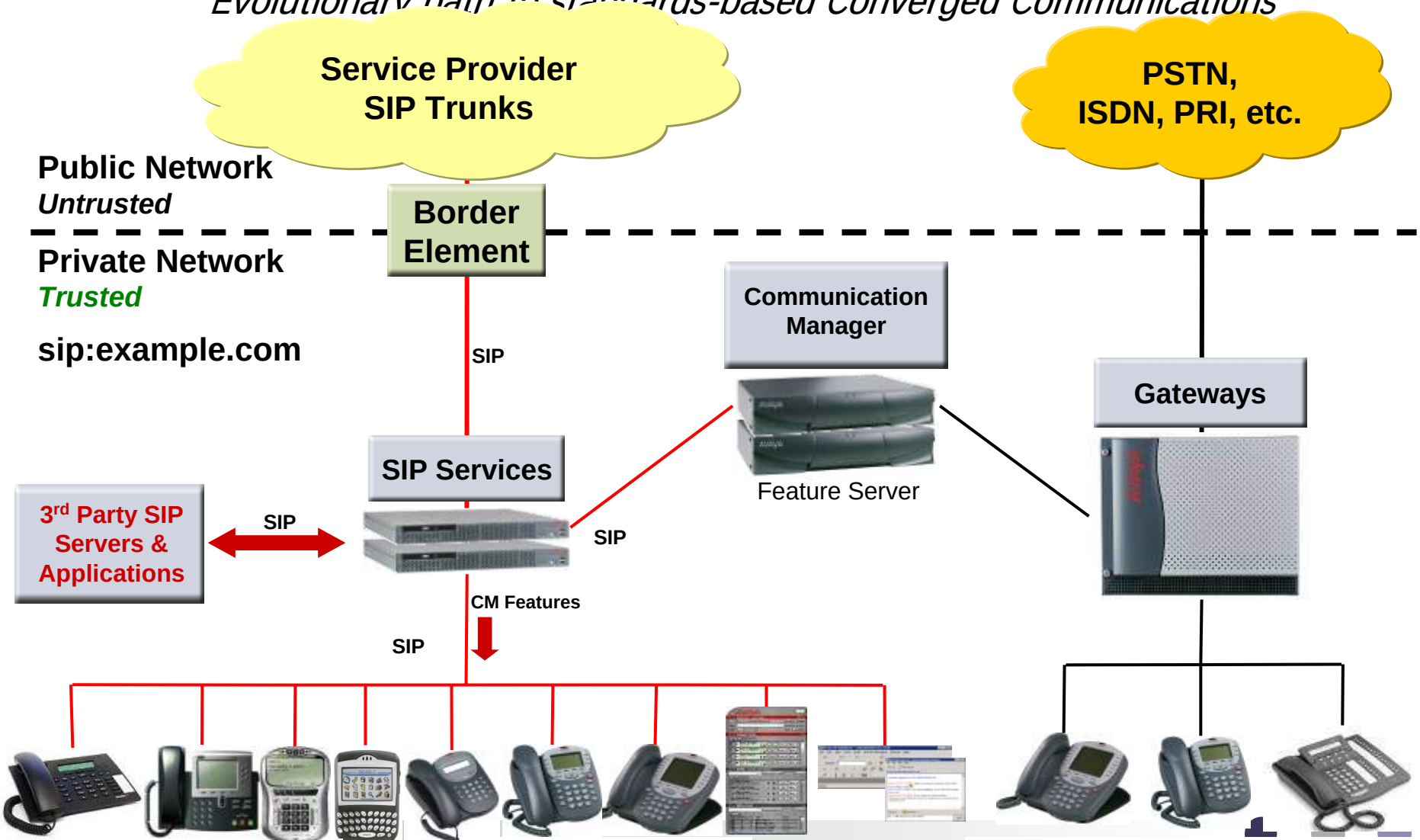
- SIP Hardware/Configurations
- Building the Solution
- SES Configuration
- CM Configuration
- Avaya Endpoints

SIP Hardware

- S8500A can be upgraded to SES 3.x.x
- S8500B can only run SES 3.x.x and later
- S8500C can only run SES 3.1.1 and later
- In a duplex configuration both servers must be the same hardware platform
- The overall system can consist of a mixture of hardware platform types
- Communication Manager used as a “Feature Server” for SIP endpoints

Avaya SIP Enablement Services

Evolutionary path to standards-based Converged Communications

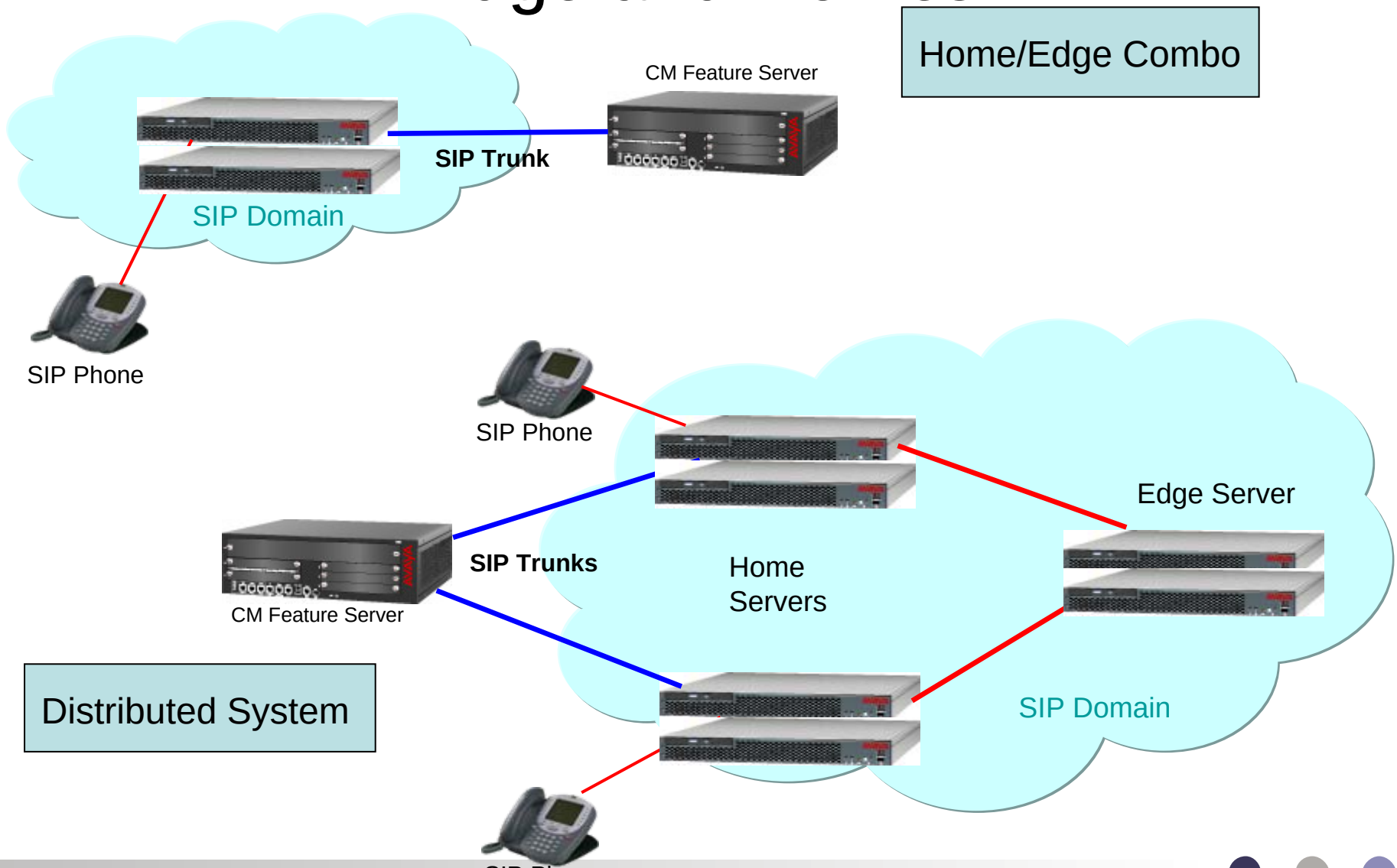


Configurations Terminology – Edge, Home, Combo

The SES can be comprised of 3 different configurations of servers

- An Edge knows about all users and which Home the users register to – speaks to all Homes and an "outbound proxy" – has "Master Administrator" privileges
  - Only 1 Edge per Domain
- The Home is where the client is registered to – speaks to the Edge, to other Homes through the Edge, and to CM
  - Up to 20 Homes per Edge
- A Combo has the functionality of both a Home and an Edge
  - Only 1 Combo in a solution
- Each of the above configurations can have a duplicated server option for redundancy

Edge and Homes



Technical – Capacities and Limitations

SES 3.1

- Requires CM 3.x
- 3,500 users per Combo or Home SES (1GB RAM)
- 6,000 users per Combo or Home SES w/ High Performance Package
  - HPP is an additional 2GB RAM (3GB total) – Edge also needs 3GB
- 1 Edge SES per domain (simplex or duplex) (3GB RAM Required)
- 20 Home SES's per administrative domain (i.e. avaya.com, example.com, etc.)
- Edge SES supports 120k BHCC (sunny day)

CM 3.1

- 16 SES instances max per CM
  - Constrained by max TLS connection limit in CM
    - Shared with other apps (Spectel, CTI, etc)
    - SES redundancy also counts towards TLS connection total
- Max SIP Trunks: S87xx = 5,000; S8700HP = 4000; S8500 = 800; S8300 = 450
  - Maximum concurrent SIP call legs
  - SIP-to-SIP call = 2 SIP call legs [SES-CM and CM-SES]
  - SIP-to-anything else (TDM, H.323, etc) = 1 SIP call leg [SES-CM]
- SIP OPS stations: Same as max station limit on CM (36k on S87xx)
  - SIP trunks are a bottleneck, limiting maximum simultaneous active SIP endpoints, limiting practical number of SIP OPS stations per CM.

Building The Solution

Building the Solution

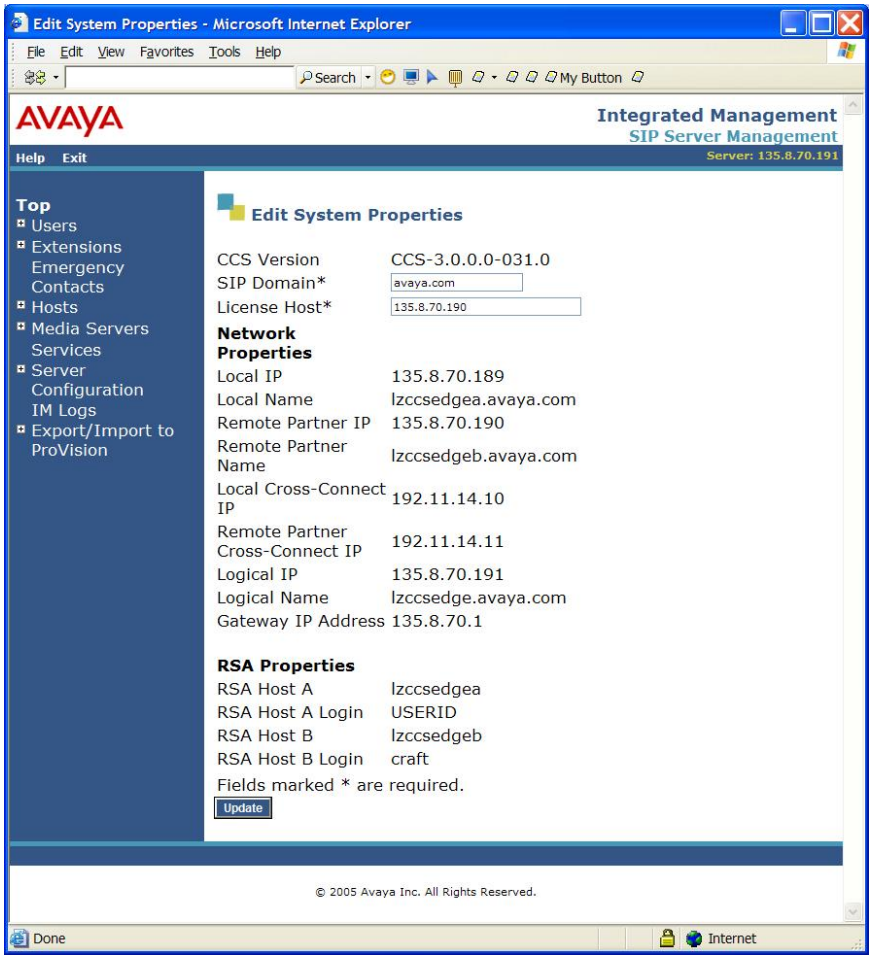
- Basic Solution Components
  - Setup and config SES
  - Setup and config CM
  - Setup and config DHCP Server (optional)
    - option-176
  - Setup and config TFTP, HTTP, HTTPS server(s) (required)
    - 46xxsettings.txt – This file is required for a 46xx SIP phone to boot and register properly.

SES Configuration

SES Administration – Key Steps

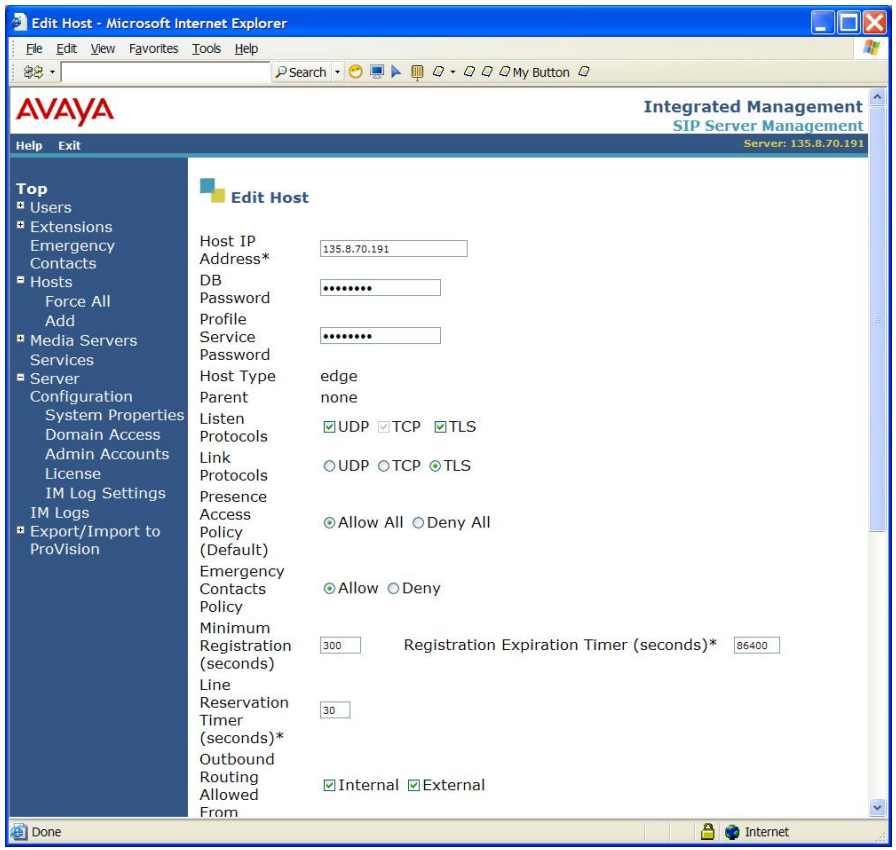
- Default Profile
  - Location demographic information
- Host
  - Host Admin
  - Host Address Map
  - Host Contact
- Media Server
  - Media Server Admin
- User
  - Handle
  - Media Server Extension
- Set time, date and timezone
- Install license and authentication files
- Schedule system backups
- Enable RSA watchdog on x305 platform
- Test INADS connectivity – ensure that on duplex servers the modem number of rings has been modified to ensure the primary server answers first

SES Admin – System Properties



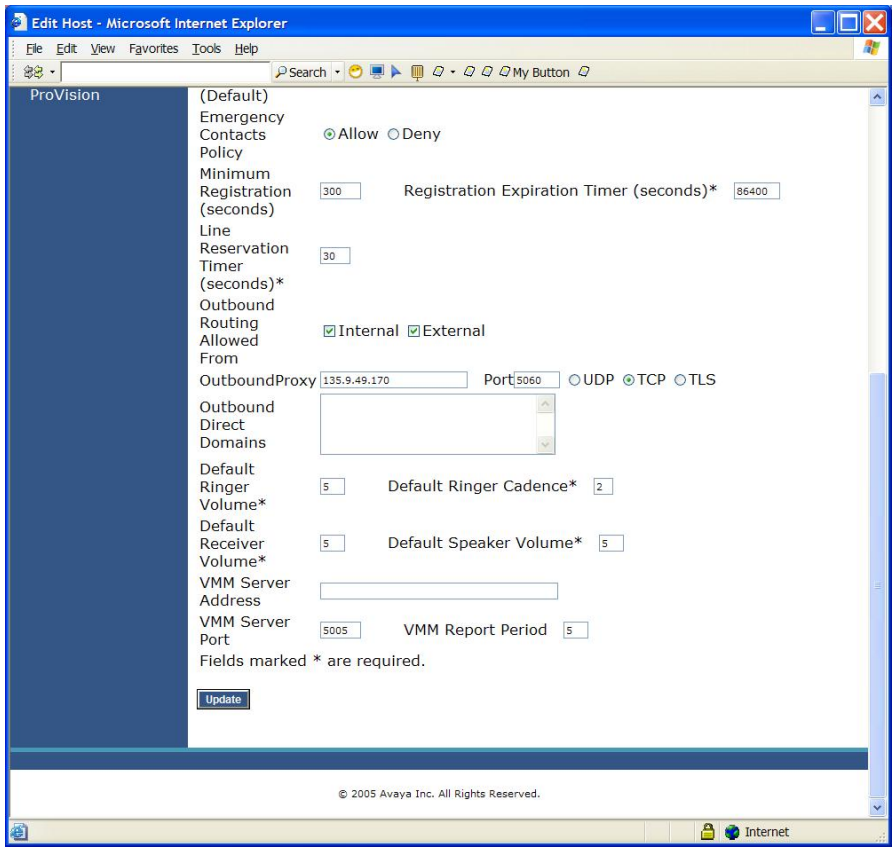
- https://<SES hostname>/admin
- Upon first opening the admin web page there will be an entry on the upper left called “Setup” – click on this to walk through configuration wizards
- SIP Domain is not the same as a DNS domain, although they can have the same syntax
- License host MUST be the physical ip address of the server the license resides on, not the logical address of the pair

Edit Host Page



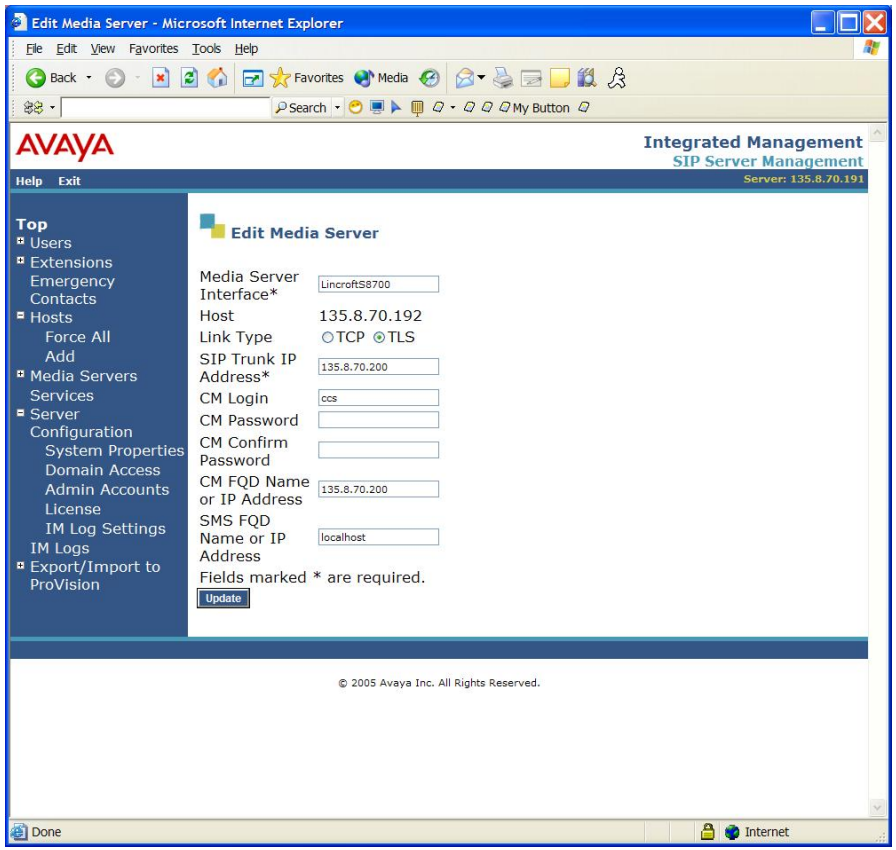
- DB Password – same DB password set during install script
- Profile Service Password – must be unique for each host – this is used for communication between “trusted” hosts – used to prevent spoofing
- Listen Protocols – Protocols used by endpoints – select all
- Link Protocols – Protocol used between SES – SES – leave set as TLS
- Presence Access Policy – default to No – change to Yes to allow Presence

Edit Host Page……cont



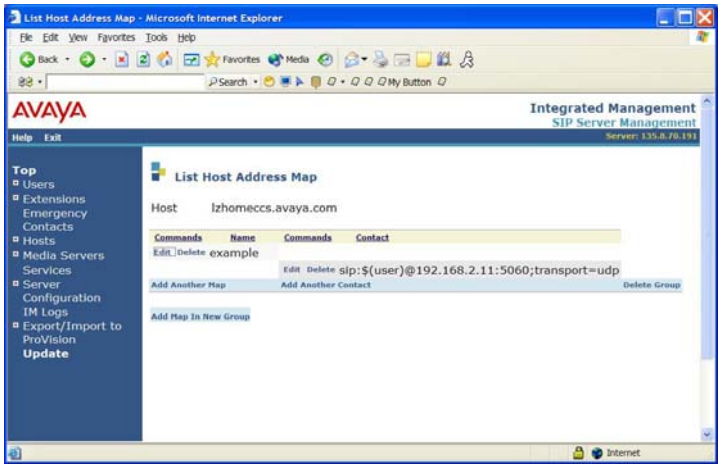
- Minimum registration and expiration timer – endpoints must use value between these when registerring
- Outbound Proxy – used as a “default” for calls going outside the domain
- Homes will set the OP to Edge and the Edge will set OP to the next device, such as a Session Border Controller
- Default Ringer, receiver, etc and VMM are specific to Toshiba endpoints (Japan only)

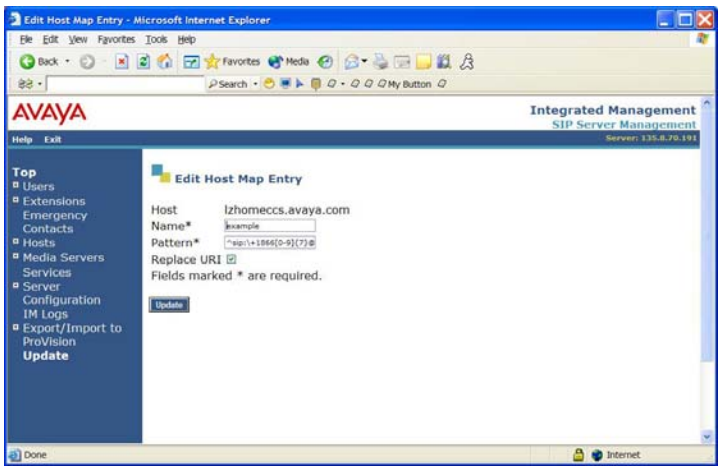
Edit Media Server



- Media Server Interface is a “friendly” name
- Link type should be TLS
- SIP Trunk IP = CLAN or procr
- CM login/password should be new static login on CM
- Password does not have **** - known issue
- CM FQD or IP – CM trunk address
- SMS FQD or IP – leave as localhost

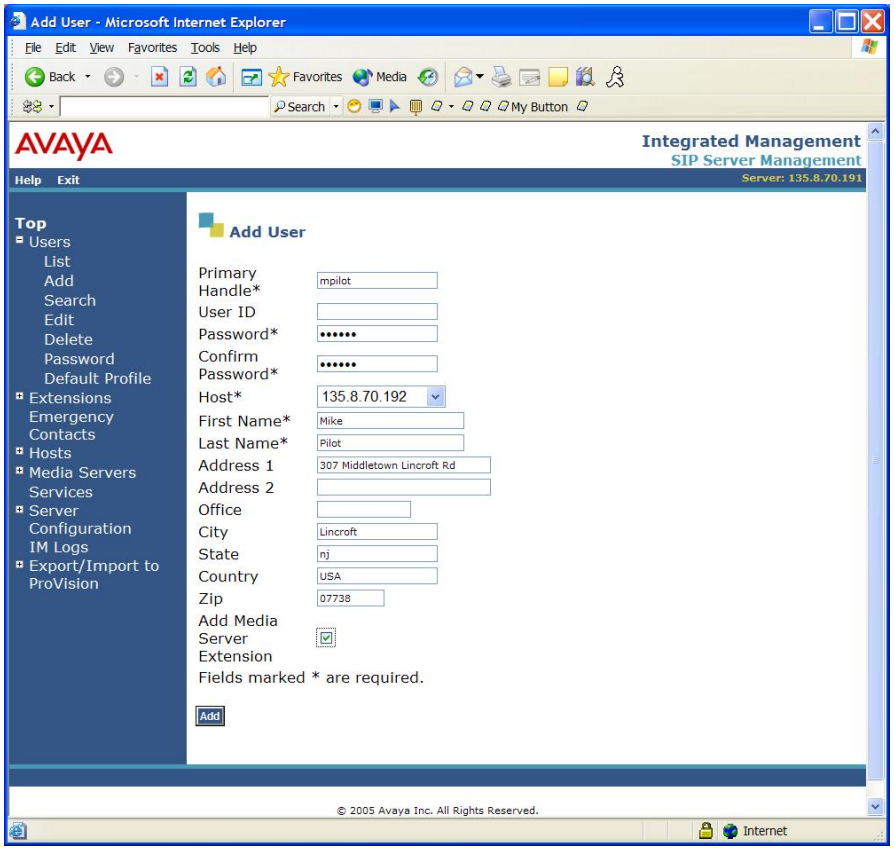
Address Maps





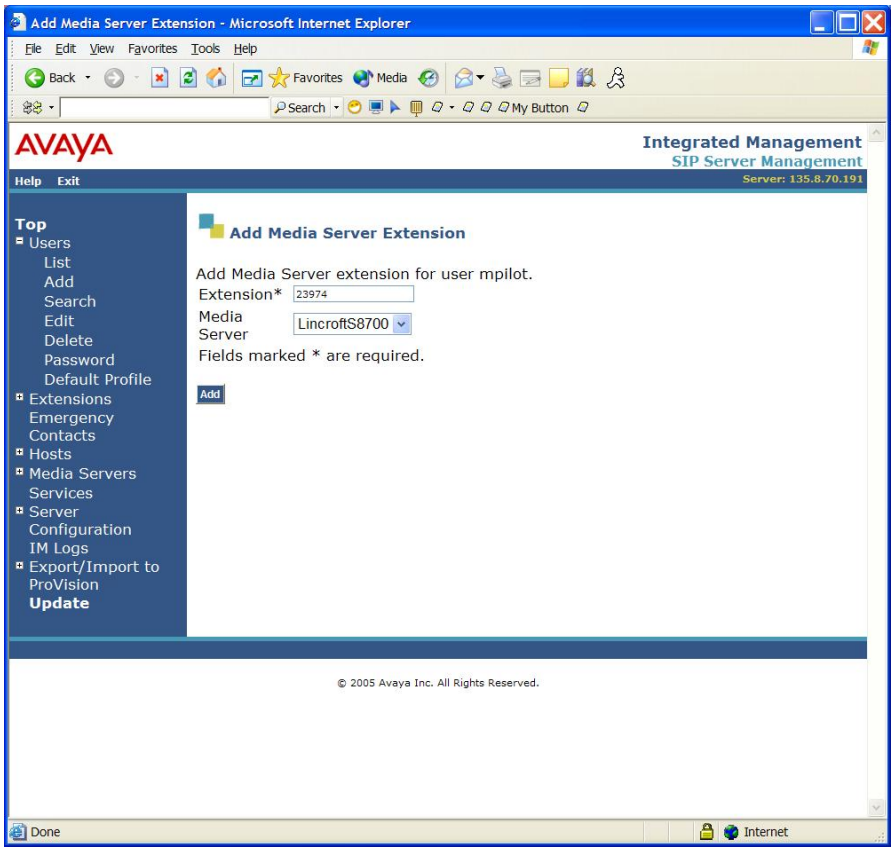
- Media Server Address Maps are not needed for OPS extensions. They are needed for non-OPS endpoints being routed from SES to CM or MM
- Host Address Maps can be used to route to 3rd party proxies for trunking to SIP Service Providers or Session Border Controllers
  - Add a contact with correct information
  - Add a map “pattern” using Linux regular expressions e.g. ^sip:5[0-9]{3}
  - Associate a contact to the map e.g. sip:$(user)@192.168.0.25:5060;transport=udp

Add a User



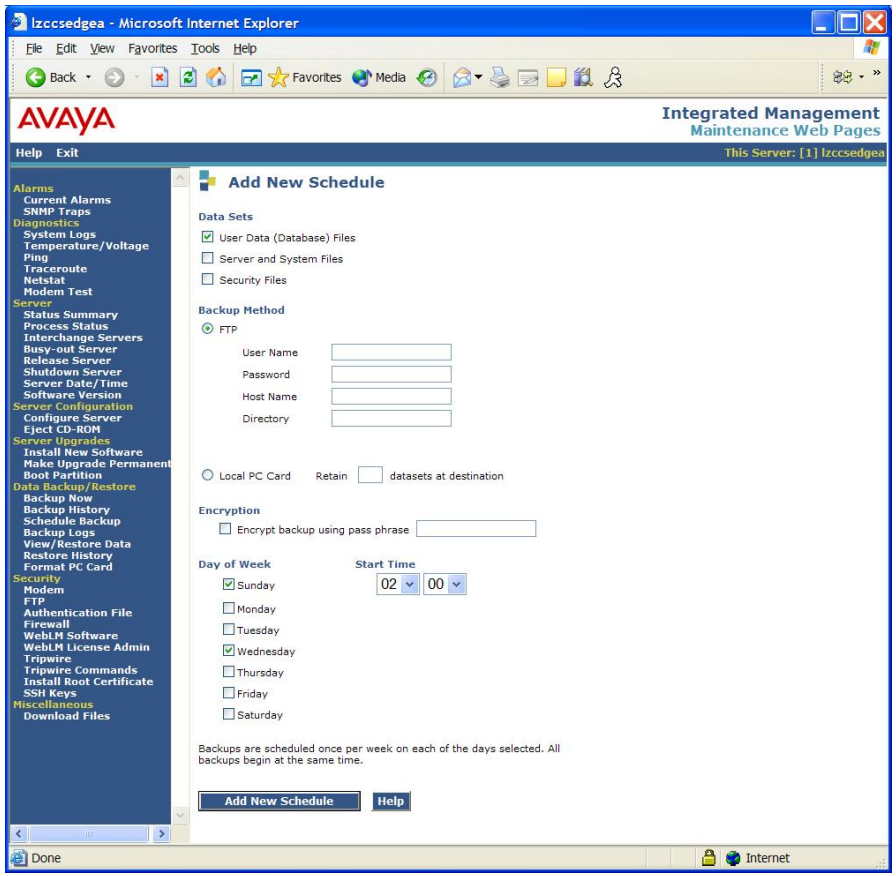
- Primary handle should be unique identifier comparable to email handle
- User ID can be left blank – it will default to Primary handle
- Password – 6 characters or more – since IP Phones can only enter numerical password, set accordingly
- SIP Softphone password can be alphanumeric
- First name, last name required
- Default Profile will populate address field, although only 1 default profile allowed
- Check Add Media Server extension to associate handle with CM station

Add a Media Server Extension



- Extension should be same as station number on CM
- Select appropriate Media Server from drop down of administered Media Servers
- Select Add
- Next screen select “continue”
- Always remember to click on **Update** to commit changes

Don’t Forget to Schedule Automated Backups!



- Under Data Backup/Restore
- Please remember to set automated backups accordingly
- Also remember to set time and date

CM Configuration

CM Administration – Key Steps

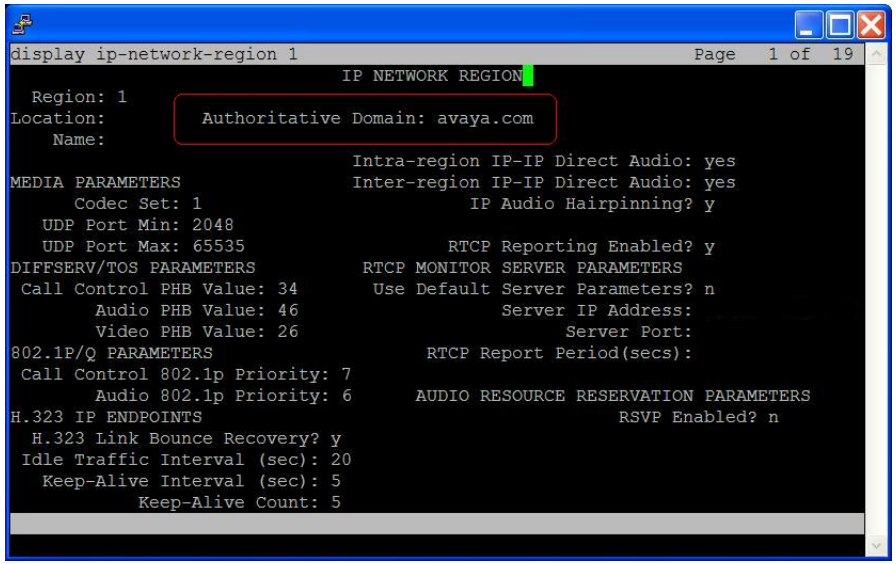
For further info see SIP Support in CM 3.0 Manual – 555-245-206

- Check RTU's – OPS, IP Trunks, SIP Trunks
- Add Node Names
  - SES Server
  - CLAN
- IP Network Region
  - Location = Location of the SES server
  - Domain = Domain of the SES server
- SIP Signaling Group
  - Group Type = SIP
  - Transport Method = TLS
  - Near End Node = CM CLAN node name
  - Far End Node = SES node name
  - Listen Ports = 5061
  - Far End Domain = SES Domain
  - DTMF over IP = rtp-payload

CM Administration – Key Steps….cont

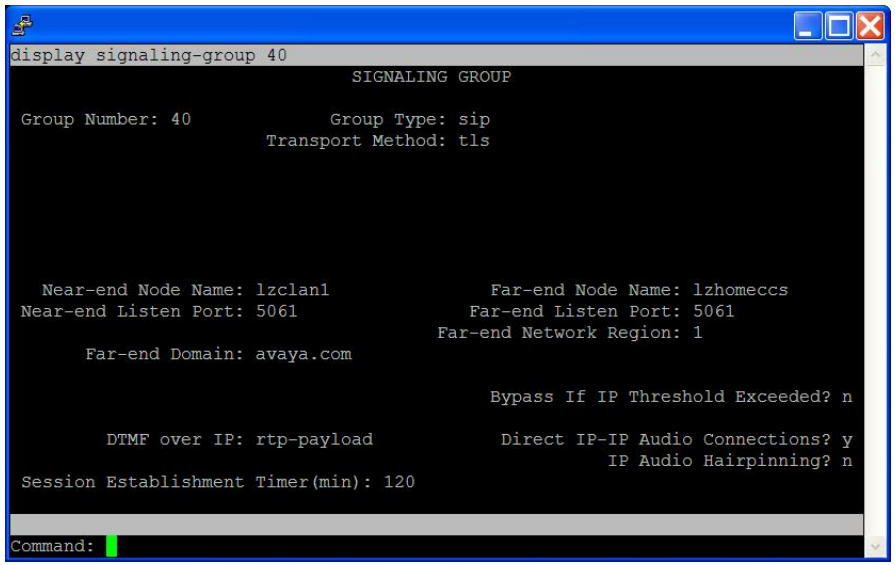
- SIP Trunk Group
  - Group Type = sip
  - Service Type = tie
- UDP/AAR
- Route Pattern
- Off Premise Station Mapping
  - Station Extension = CM OPS extension number
  - Application = OPS
  - Phone Number = SES Media Server extension number
  - Trunk Selection = SIP trunk, AAR, or ARS
  - Configuration Set (if applicable) = SIP specific config set
- Ensure MedPro and CLAN are running the latest firmware

IP Network Region



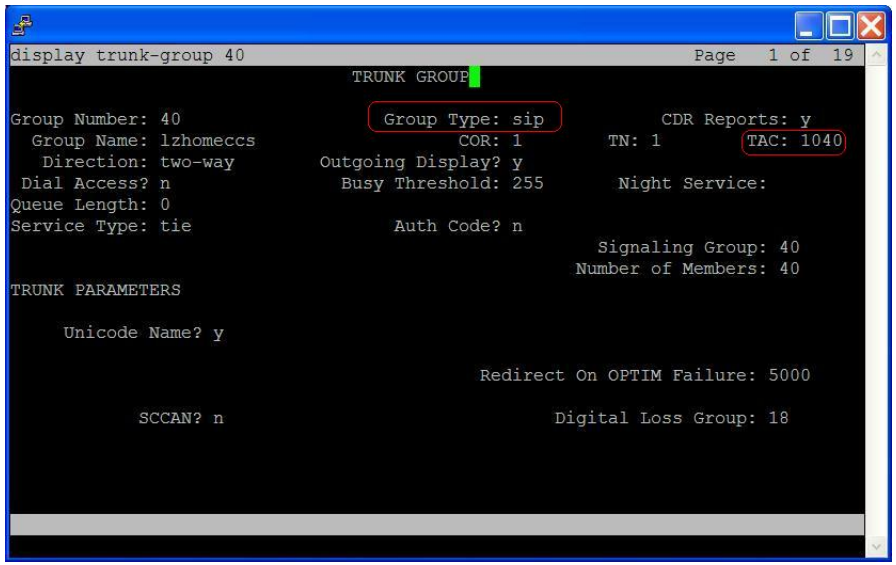
- Authoritative Domain must be set to match the SIP domain in use in the solution – THIS IS THE MOST COMMONLY MISSED ADMINISTRATION STEP. CALLS WILL NOT WORK UNLESS THIS IS CORRECT

Signalling Group



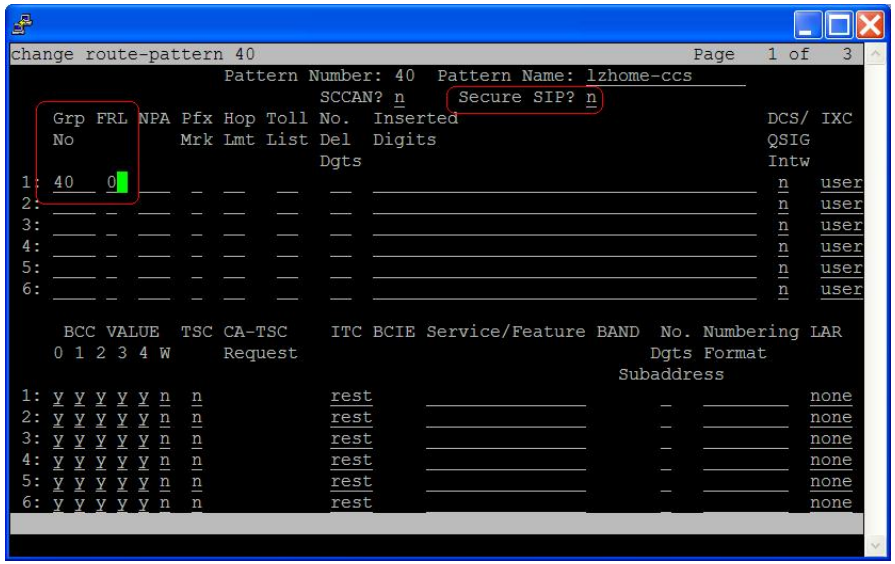
- Group Type = sip – sets format of page
- Transport method = tls
- Near-end/Far-end Node names – need to have been administered on node-names page – ports = 5061
- Far end domain = SIP domain on SES
- DTMF over IP = rtp-payload

Trunk Group



- Set group type = sip
- TAC needs to be setup based upon dialplan analysis table – same as usual (Dial Access is not available)
  - "list trace tac nnn" is useful for troubleshooting
- Service Type = tie
- ROOF – defaults to 5000
  - 5 seconds until calling party will hear failure tone

Route Pattern


change route-pattern 40 Page 1 of 3
Pattern Number: 40 Pattern Name: lzhome-ccs
SCCAN? n Secure SIP? n
Grp No FRL NPA Pfx Mrk Hop Lmt Toll List No. Del Dgts Inserted Digits DCS/ QSIG Intw IXC
1: 40 0 n user
2: n user
3: n user
4: n user
5: n user
6: n user
BCC VALUE 0 1 2 3 4 W TSC CA-TSC Request ITC BCIE Service/Feature BAND No. Dgts Subaddress Numbering Format LAR
1: y y y y y n n rest none
2: y y y y y n n rest none
3: y y y y y n n rest none
4: y y y y y n n rest none
5: y y y y y n n rest none
6: y y y y y n n rest none

- Once UDP/AAR info has been administered follow standard Route Pattern setup
- Secure SIP should be set no - default, only set to yes if...
  - end to end conversation is using tls including endpoints

    OR
  - A SIP Softphone only solution and only if using tls
- Configurable option for SCCAN
  - If this is set to yes, Avaya IP Phones will not operate as expected

Business-Class SIP Telephony Features

Available to Any SIP Telephone

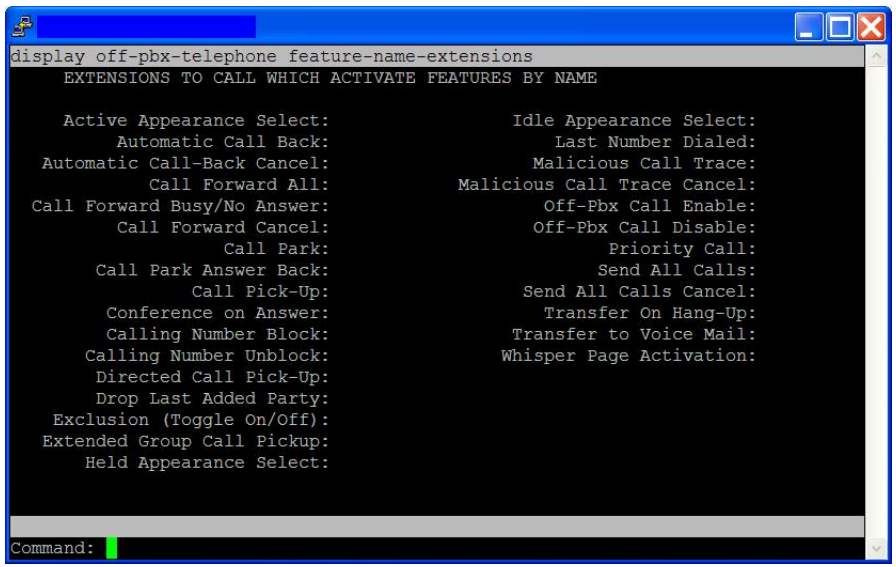
Station Side Features

1. Active Appearance Select
2. Automatic Call Back
3. Call Forwarding – All
4. Call Forwarding – Busy
5. Call Forwarding – No Answer
6. Call Forwarding Deactivation
7. Call Hold
8. Call Park
9. Call Park Answer Back
10. Group Call Pick-Up
11. Calling Party Number Block/Unblock
12. Conference on Answer
13. Consultation Hold
14. Directed Call Pick-Up
15. Distinctive Alerting
16. Drop Last Added Party
17. Exclusion
18. Extended Group Call Pick-Up
19. Group Paging
20. Held Appearance Select
21. Idle Appearance Select
22. Last Number Dialed
23. Malicious Call Trace Activation/Deactivation
24. Message Waiting Indication
25. Multiple Call Handling
26. Priority Call
27. Send All Calls
28. Transfer – Attended
29. Transfer – Unattended
30. Transfer on Hang Up
31. Transfer to Voice Mail
32. Auto-Intercom
33. Dial Intercom

Trunk Side Features

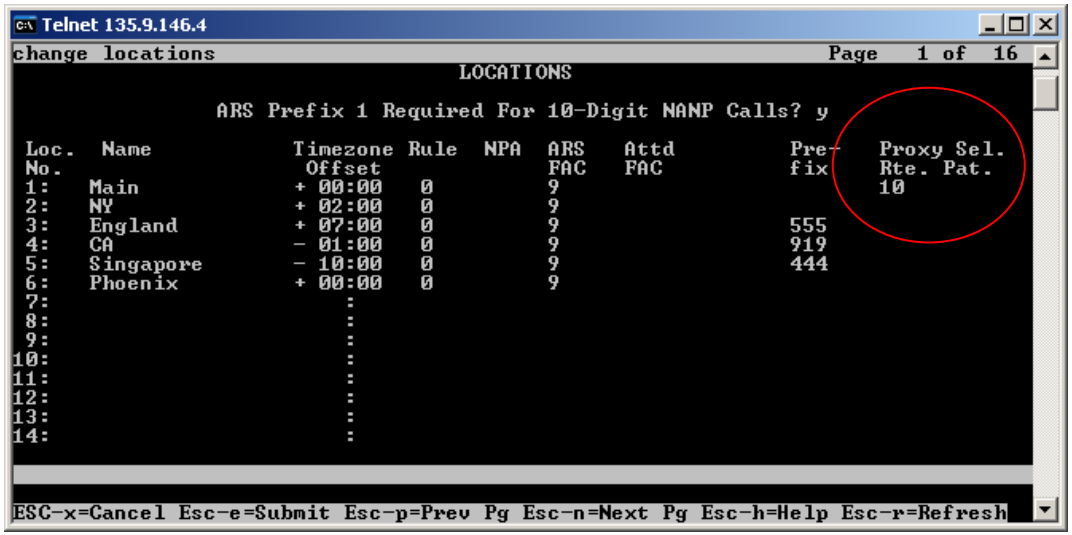
32. Automatic Alternate Routing
33. Automatic Route Selection
34. Announcements
35. Auto Answer Intercom
36. Automatic Call Distribution (ACD)
37. Bridged Appearances
38. Call Detail Records (CDR)
39. Centralized Attendant Service
40. Class of Restriction (Call Screening)
41. Class of Service
42. Codec Preferences
43. Crisis Alert to Digital Pager
44. Controlled Toll Restrictions
45. Dial Plan Expansion
46. Enhanced 911/CAMA Trunk Interface
47. Extension to Cellular (EC500)
48. Find-Me
49. Hospitality – Controlled Restriction
50. IP Traffic Measurements
51. Manual Signaling
52. Meet-Me Conferencing
53. Music on Hold
54. Night Service
55. Outgoing Trunk Queuing
56. Service Observing
57. 3-way Conference – 3rd Party Added
58. Transfer Recall
59. Trunk Group Hunting
60. Time of Day Routing
61. Uniform Dial Plan
62. Vectoring

Off-pbx-telephone feature-name-extensions



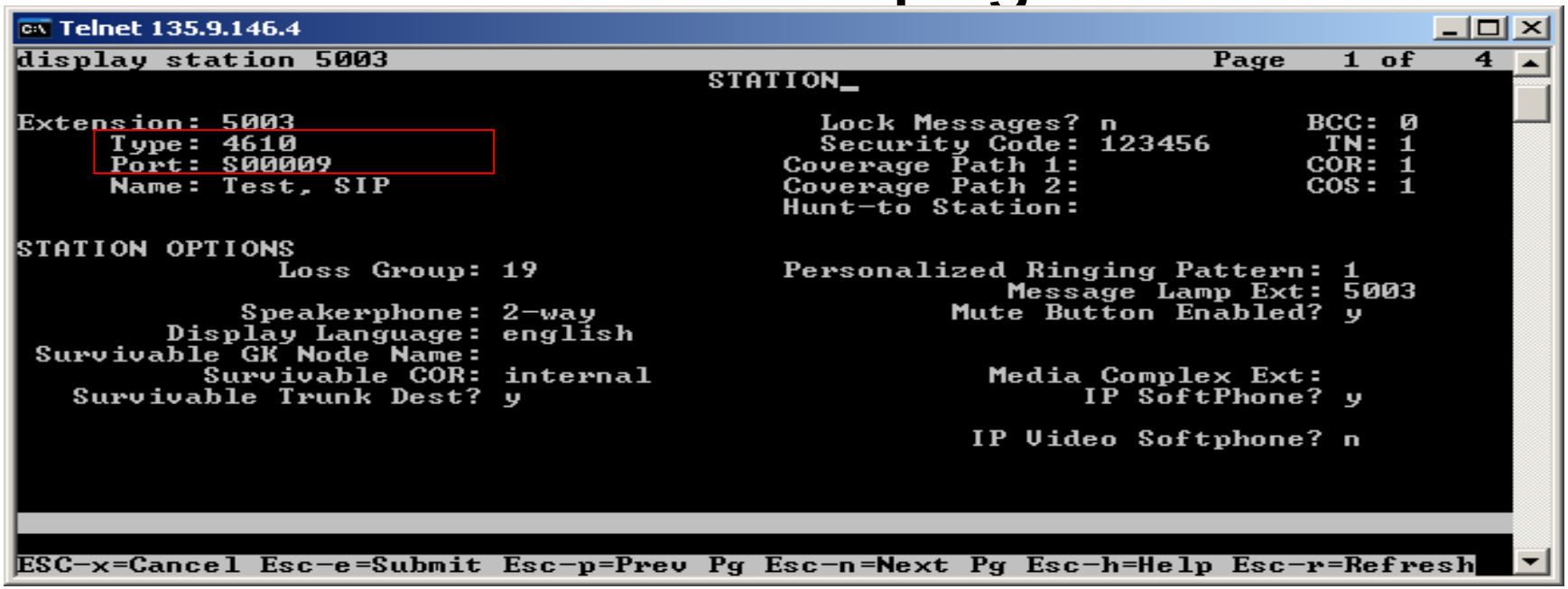
- These CM features can be integrated into all SIP endpoints in the solution
- Configure valid dialable station numbers to each feature
- On the SIP endpoint, dial the station for the feature you would like to activate
- Add individually desired features as Speed Dials on the phone with “friendly names”
  - i.e. SAC on, SAC off

Locations


Telnet 135.9.146.4
change locations
Page 1 of 16
LOCATIONS
ARS Prefix 1 Required For 10-Digit NANP Calls? y
Loc. No. | Name | Timezone Offset | Rule | NPA | ARS FAC | Attd FAC | Prefix | Proxy Sel. Rte. Pat.
1: Main + 00:00 0 9 10
2: NY + 02:00 0 9
3: England + 07:00 0 9 555
4: CA - 01:00 0 9 919
5: Singapore - 10:00 0 9 444
6: Phoenix + 00:00 0 9
7:
8:
9:
10:
11:
12:
13:
14:
ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh

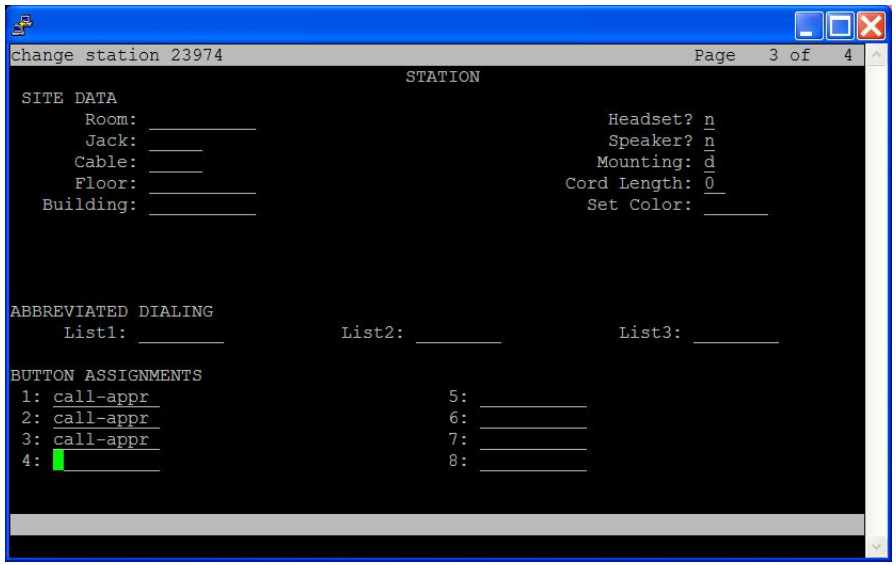
- Proxy Sel. Route Pattern = SIP Trunk Group Route Pattern number.
  - This is required when you are routing calls to a domain that is not administered on the ip network region form.
  - This is required for CM to resolve alphanumberic sip URIs.

Station form page 1



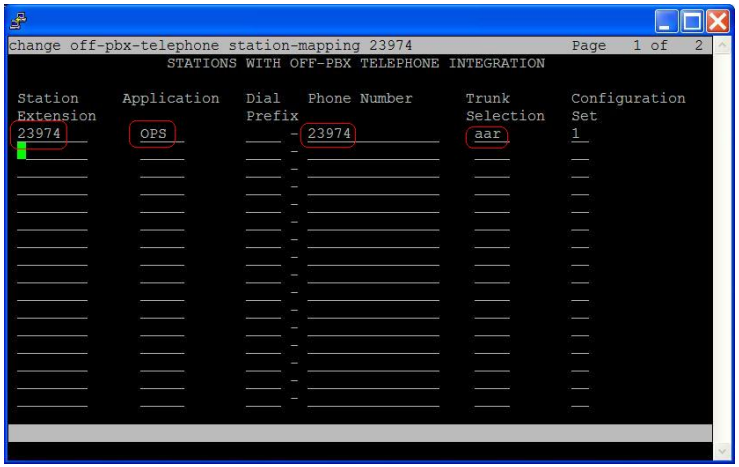
- Type should be any 46XX IP
- A DCP station type should NOT be used for a SIP OPS station. If bridging is used for that station and the CM system has TTI enabled corruption can occur.
- Message Lamp needs to be correct

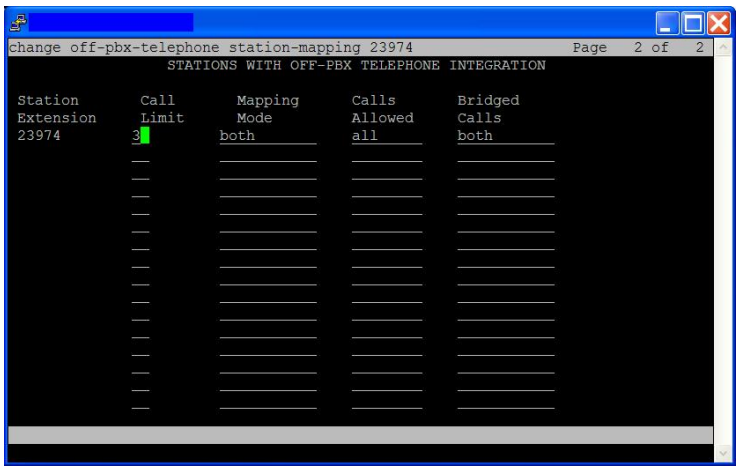
Station form page 3



- Accept the default of 3 Call-Appr
  - If being used for SIP Softphone the number of apperances should be set to 5.
- No other changes need to be made
- Advanced SIP Telephony Features require administration of each used feature as a button on the OPS station form.

off-pbx-telephone station-mapping



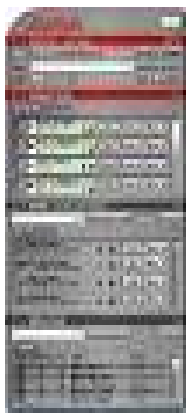


- cha off sta nnnnn
- Station Extension from previous slide
- Application = OPS
- Phone number = media server extension on SES admin – should be the same as station extension
- Trunk selection can be aar, ars, or trunk id. The actual trunk id number *could* be specified – helps for troubleshooting
- Page 2 – set call limit to be same as Session Appearances on Phone – default is 2 – change to 3 and set Bridged calls to none
- Configuration set defaults are fine

Avaya Endpoints

Avaya Enpoints

- Avaya SIP Enabled Endpoints
  - 4602, 4610, 4620, and 4621
  - Avaya SIP Softphone
  - Avaya IP Softphone v5.2
  - Avaya IP Agent v6



46xx SIP Phones

- Avaya R2.2.2 SIP Phones
  - 4602, 4610, 4620, 4621 all supported with SIP firmware version 2.2.2 Equivalent user applications available (e.g. Call Log, Speed Dial, Web) for 4602/10/20/21SW
    - Refer to the 4602 R 1.1 to 4602 2.2.2 Conversion Job Aid on Avaya Support or the Anatomy of a Successful Cut websites.
    - One-X endpoints will support the SIP protocol in the September release.
  - Administration is done via the 46xxsettings.txt file and not an individual phone web interface
- Hardware
  - Same phone for H.323 and SIP
  - H.323 is the default protocol from the factory
  - Protocol (H.323 or SIP) is selected from the keypad (MUTE 744 #) or by the type of 46xxsettings.txt file
- TFTP, HTTP, and HTTPS are supported for file downloads
- CODECS supported
  - G.711 mu/a (All Avaya SIP Phones)
  - G.729B (SIP Softphone Only)
  - G.729A (46xx SIP phones Only)

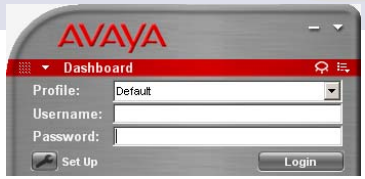
Dial Plan example

- **“[2-4]xxx|[68]xxx|9Zxxxxxxxxxx”**
- **[2-4]xxx**: Four-digit dial extensions, with valid extensions starting with 2, 3, or 4;
- **[68]xxx**: Four-digit dial extensions, with valid extensions starting with 6 or 8;
- **9Z11xxxxxxxxx**: Network Access Code (“9 for an outside line”), followed by dial tone, followed by any string of 11 digits– typical instance of Automatic Route Selection (ARS) in the US

Avaya SIP Softphone R2.1

Overview Description

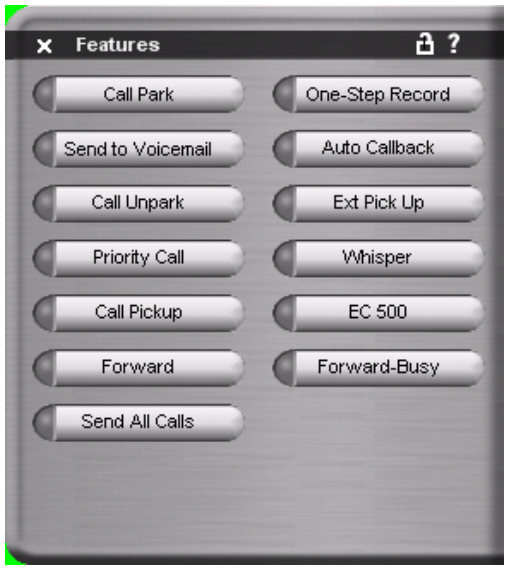
Avaya SIP Softphone R2.1

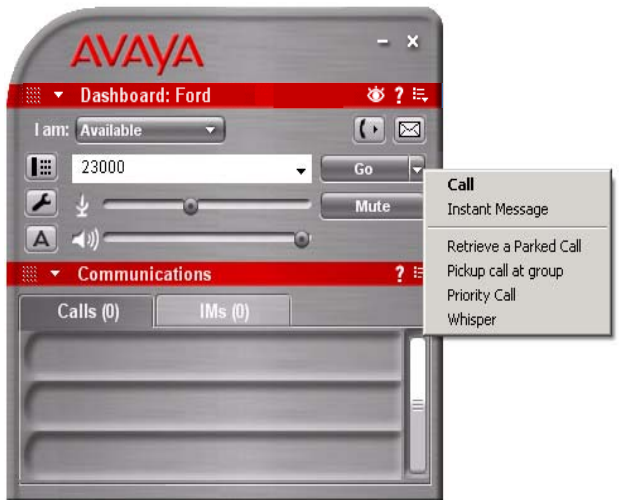


- Avaya SIP Softphone is a generic SIP endpoint with extras
  - Enhanced Conferencing features
  - Additional CM-based features such as Priority Call, Call Forwarding, and other AST features described in slide 78
  - Implements SIP for telephony, IM, and presence
  - Network-based contact store and access control lists via SES/PPM
  - Desktop integration with Microsoft Outlook, Lotus Notes, LDAP, and Microsoft Internet Explorer
  - Additional Desktop integration capabilities with Microsoft Smart Tags
  - Bluetooth Integration
- Continues to expand upon the user interface introduced in SIP Softphone R2
- Supports Road Warrior configuration only
  - No Telecommuter
  - No Shared Control
- Can discover configuration settings (by accessing 46xxsettings.txt file via HTTP)

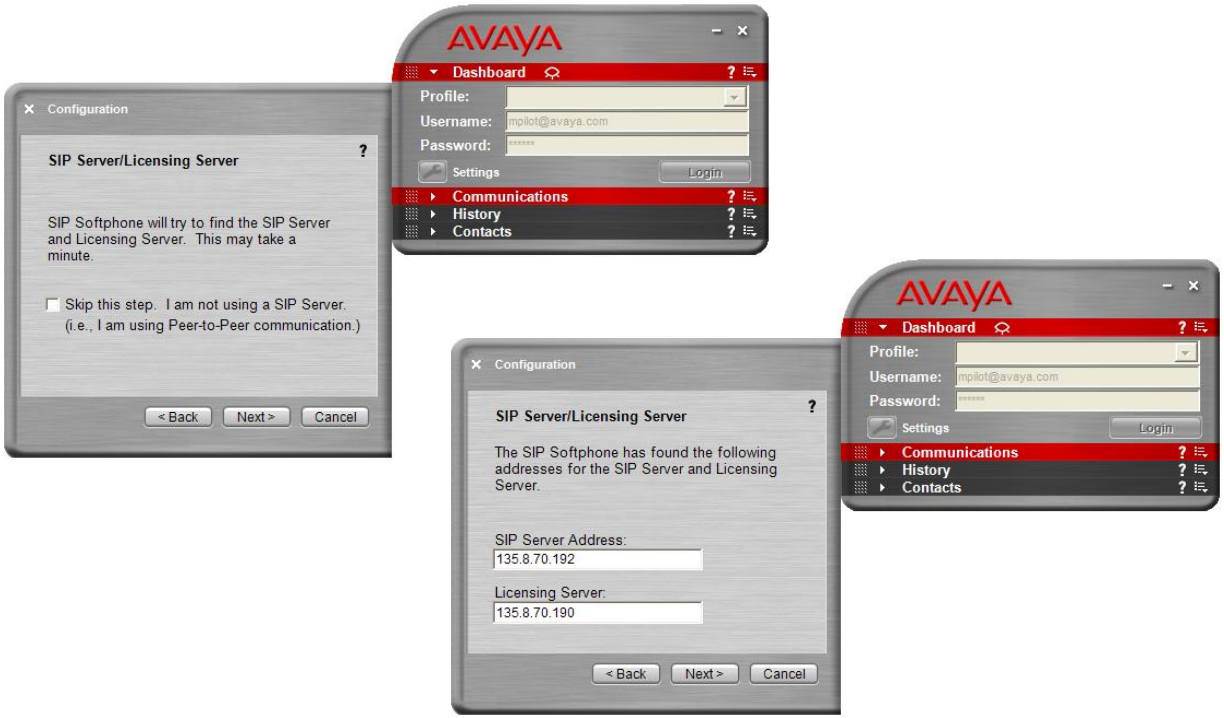
Avaya SIP Softphone – User Interface (cont.)

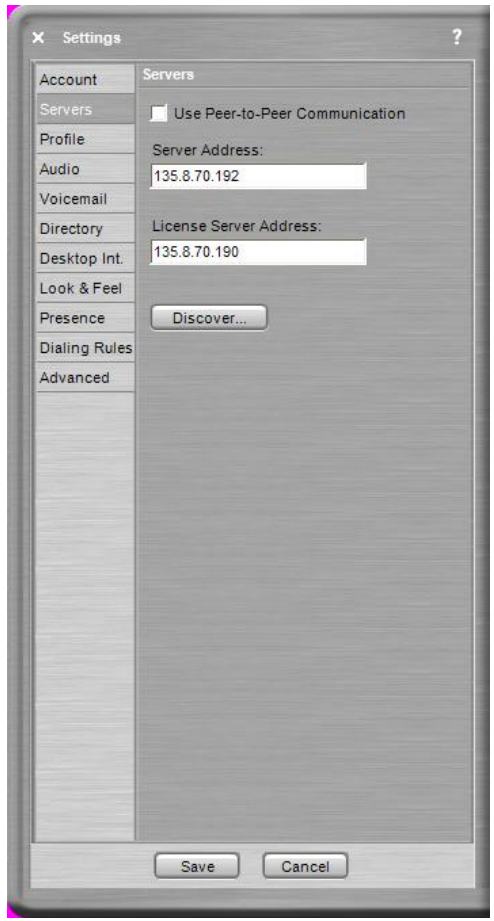
- Features are provisioned in Communication Manager
- Feature Panel lists available features





Proxy and License Server



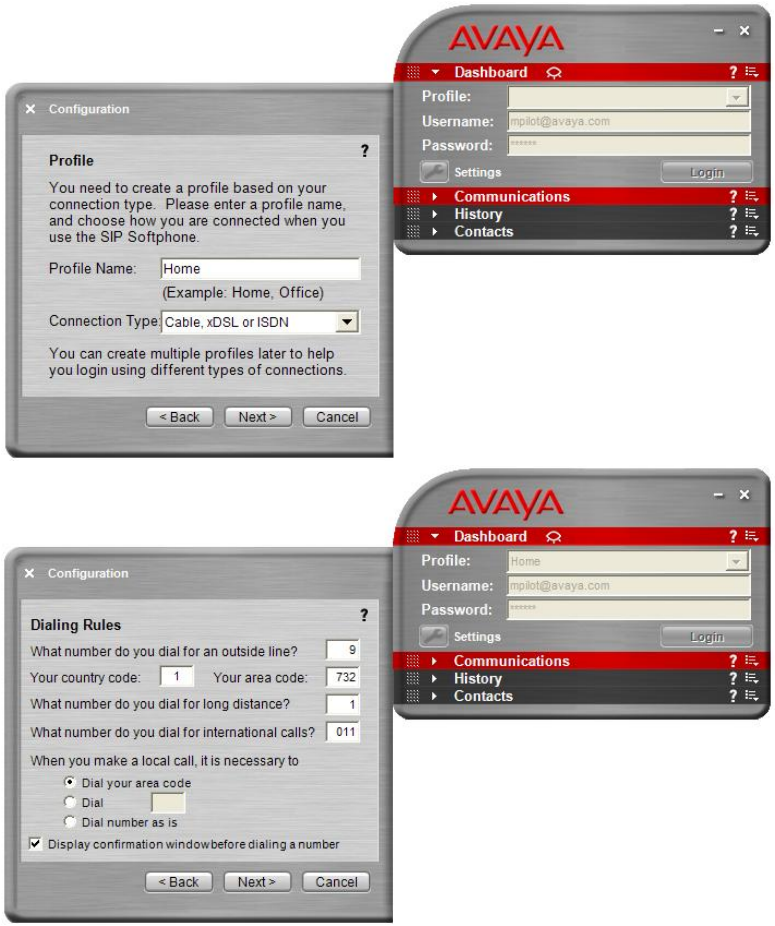


- SIP Softphone will try to find a Proxy and license server through DHCP settings along with other parameters
- After initial config, the “Discover” feature is available from Settings -> Server -> Discover

SIP Softphone and 46xxsettings.txt

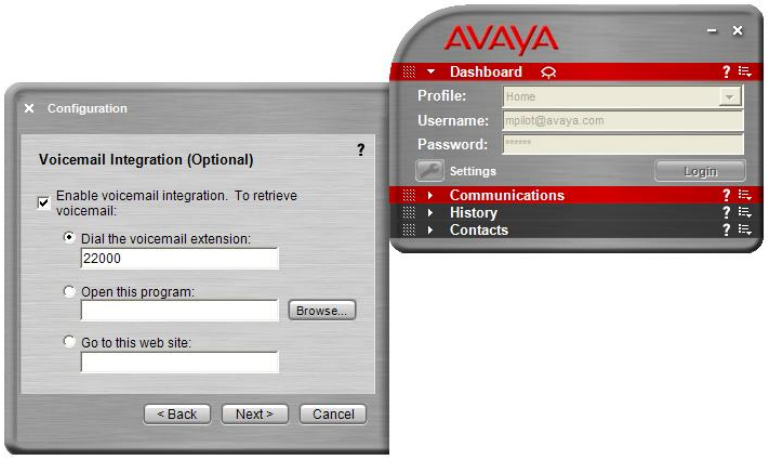
- During initial configuration, SIP Softphone will look for Option 176 in the DHCP scope
- SIP Softphone can obtain information about the SIP server address, license server address, and LDAP directory by using DHCP (and the “Discover function) to obtain the 46xxsetting.txt file used by the IP Telephones. SIP Softphone will use the HTTP server address provided in DHCP OPTION 176 to find the 46xxsettings.txt file. SIP Softphone will then read the following values from that file:
  - SIPPROXYSRVR - This is the address of the SIP Proxy/Registrar
  - WEBLMSRVR - This is the address of the Licensing server.
  - SP_DIRSRVR - This is the address of the LDAP server.
  - SP_DIRSRVRPORT - This is the port of the LDAP server.
  - SP_DIRTOPDN - This is the search root of the LDAP server.

Config continued



- Setup Profiles – select the Profile you wish to use when you login
- Profile settings are similar to “Bandwidth configuration” from IP Softphone
- Used for CODEC advertisement, not the physical connection
  - Connection Type
    - LAN – 711mu/a, 729a, and 723
    - Cable – 729a and 723
    - 28,800 or faster – 723 only
- Dialing Rules

Voice Mail Config



- Check the box for Enable voicemail integration
- Enables SIP Softphone to perform a function upon clicking on the voicemail icon while registered
- Envelope turns bright red when a voice mail is waiting to be picked up